FXOA GATEWAY
In SIP
User Manual
(2/4 Ports)

STEPS IN CONFIGURATION 4

1. START UP 5

1.1 INTRODUCTION 5
1.2 FEATURES AND SPECIFICATION 6
1.3 ACCESSORIES AND EQUIPMENT 8
1.4 APPEARANCE 9

2. HOW TO SETUP AND CONNECT BASICALLY 12

2.1 SYSTEM REQUIREMENT 12
2.2 IP ENVIRONMENT SETTING 13
2.2.1 For Windows 2000/NT 13
2.3 NETWORK CONFIGURATIONS IN YOUR GATEWAY 15
2.3.1 Static ip address 17
2.3.2 DHCP mode 19
2.3.3 PPPoE mode 20
2.4 MAKING A VOIP CALL 20
2.4.1 Configure the gateway into the Peer-to-Peer mode 21
2.4.2 Configure the gateway into the Proxy mode 24

3. ADVANCE CONFIGURATIONS 27

3.1 NETWORK INTERFACE 27
3.2 SIP INFORMATION 29
3.3 SECURITY CONFIG 30
3.4 LINE CONFIGURATION 31
3.5 SYSTEM CONFIGURATION 32
3.6 VOICE CONFIGURATION 35
3.7 TONE SETTING 36
3.8 PHONE BOOK 37
3.9 PREFIX CONFIGURATION 38
3.10 ROUTING TABLE 41
3.11 FXO PASSWORD 43
3.12 IP PACKET TOS 44
3.12 PASSWORD CONFIGURATION 44
3.13 ROM UPGRADE 45
3.14 FLASH CLEAN 49
3.15 COMMIT 49
3.16 REBOOT SYSTEM 50

4. COMMAND LIST 51

4.1 HYPER TERMINAL SETTING 51
4.2 COMMAND LIST 54
4.2.1 [help] *54*
4.2.2 [quit] *55*
4.2.3 [debug] *55*
4.2.4 [reboot] *55*
4.2.5 [flash] *56*
4.2.6 [commit] *56*
4.2.7 [ifaddr] *57*
4.2.8 [time] *58*
4.2.9 [ping] *58*
4.2.10 [sysconf] *59*
4.2.11 [sip] *62*
4.2.12 [security] *65*
4.2.13 [line] *65*
4.2.14 [route] *67*
4.2.15 [prefix] *69*
4.2.16 [pause] *70*
4.2.17 [pbook] *71*
4.2.18 [voice] *73*
4.2.19 [tone] *75*
4.2.20 [fxopwd] *77*
4.2.21 [record] *77*
4.2.22 [support] *85*
4.2.23 [tos] *85*
4.2.24 [pt] *86*
4.2.25 [rom] *87*
4.2.26 [passwd] *88*
4.2.27 [auth] *89*

STEPS IN CONFIGURATION **4**

1. START UP **5**

1.1 INTRODUCTION 5
1.2 FEATURES AND SPECIFICATION 6
1.3 ACCESSORIES AND EQUIPMENT 8
1.4 APPEARANCE 9

2. HOW TO SETUP AND CONNECT BASICALLY **12**

2.1 SYSTEM REQUIREMENT 12
2.2 IP ENVIRONMENT SETTING 13
2.2.1 For Windows 2000/NT *13*
2.3 NETWORK CONFIGURATIONS IN YOUR GATEWAY 15
2.3.1 Static ip address *17*
2.3.2 DHCP mode *19*
2.3.3 PPPoE mode *20*
2.4 MAKING A VOIP CALL 20
2.4.1 Configure the gateway into the Peer-to-Peer mode *21*
2.4.2 Configure the gateway into the Proxy mode *24*
2.4.3 Configure the gateway into the Gateway mode *25*

3. ADVANCE CONFIGURATIONS **27**

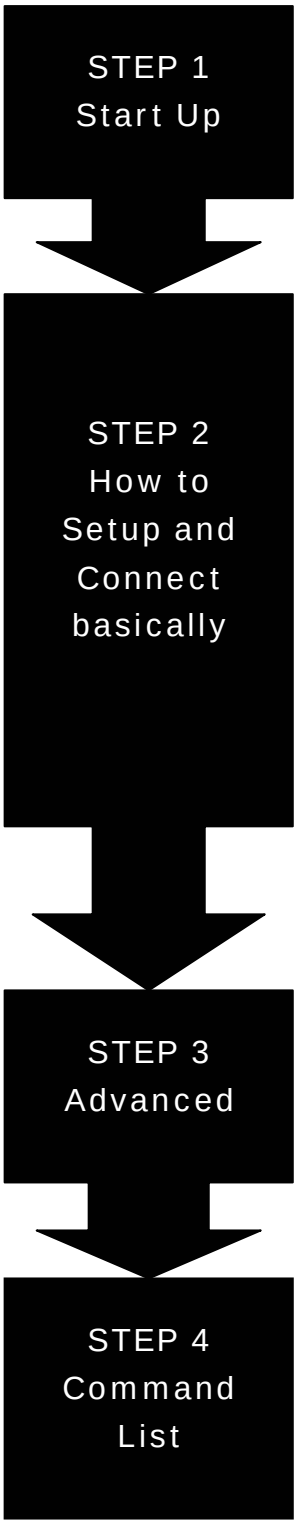
3.1 NETWORK INTERFACE 27
3.2 SIP INFORMATION 29
3.3 SECURITY CONFIG 30
3.4 LINE CONFIGURATION 31
3.5 SYSTEM CONFIGURATION 32
3.6 VOICE CONFIGURATION 35
3.7 TONE SETTING 36
3.8 PHONE BOOK 37
3.9 PREFIX CONFIGURATION 38
3.10 PAUSE CONFIGURATION 41
3.11 ROUTING TABLE 41
3.12 FXO PASSWORD 43
3.12 IP PACKET TOS 44
3.13 IP PACKET TOS 44
3.14 PASSWORD CONFIGURATION 44
3.15 RTP PAYLOAD TYPE CONFIGURATION 45

3.16 ROM UPGRADE 46
3.17 VERSION INFORMATION 49
3.18 FLASH CLEAN 49
3.19 COMMIT 49
3.20 REBOOT SYSTEM 50

4. COMMAND LIST 51

4.1 HYPER TERMINAL SETTING 51
4.2 COMMAND LIST 54
4.2.1 [help] 54
4.2.2 [quit] 55
4.2.3 [debug] 55
4.2.4 [reboot] 55
4.2.5 [flash] 56
4.2.6 [commit] 56
4.2.7 [ifaddr] 57
4.2.8 [time] 58
4.2.9 [ping] 58
4.2.10 [sysconf] 59
4.2.11 [sip] 62
4.2.12 [security] 65
4.2.13 [line] 65
4.2.14 [route] 67
4.2.15 [prefix] 69
4.2.16 [pause] 70
4.2.17 [pbook] 71
4.2.18 [voice] 73
4.2.19 [tone] 75
4.2.20 [fxopwd] 77
4.2.21 [record] 77
4.2.22 [support] 85
4.2.23 [tos] 85
4.2.24 [pt] 86
4.2.25 [rom] 87
4.2.26 [passwd] 88
4.2.27 [auth] 89

Steps in configuration



To check out the peripheral equipments and understand the feature of this gateway. Please read this step very carefully before starting the configuring.

Connecting the gateway and computer to start configuring by WEB GUI.

Setting the ip address for this gateway to make sure that it could connect with the internet.

Setting the configurations of dialing, including the Peer-To-Peer, GK mode and how to set these tables to make calls by this gateway easily.
The other configurations of make call will be discussed in this step.

Advanced configurations and special functions of this gateway. Using the WEB GUI to show how to set this table and explain the meaning of these tables.

To explain the meaning of the command in the command line interface and example the usage of the command.
To get more usages or configuration in this step and study about the command line configuration.

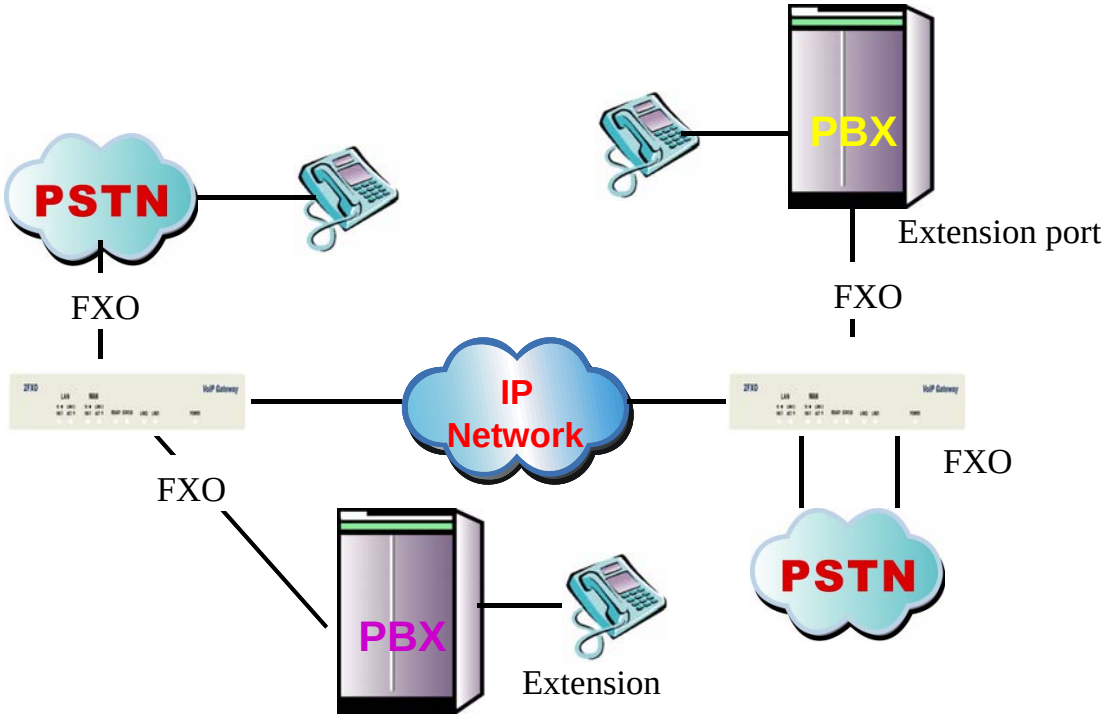
1. Start Up

1.1 Introduction

The FXO gateway provides voice/fax service over IP network with SIP protocol. By connecting to your existing ADSL or cable modem service, which allows the use of a single, network for voice and fax services with consequent saving in network infrastructure and greatly reduced telephone charges. Ideal solution for providing low cost communications between headquarters and branch offices in the world, as well as for SOHO and office telephony applications.

FXO Gateways provides analog lines to connect local PSTN/PTT interface (FXO), and converts voice/fax signal onto IP network. The management feature is via RS-232C COM port and TELNET.

Application Architecture



- ***FXO ports can connect with PSTN Line or Extension Line of PBX***

1.2 Features and specification

Features

- IETF RFC 3261
- Automatically Dial Path Selection (IP or PSTN)
- PSTN Line switch to telephone set when power is failure
- PPPoE support
- Behind NAT router or IP sharing device
- DNS server inquiry
- Provide Peer-to-Peer Mode (Non SIP Proxy needed) selection
- E.164 Dial Plan
- TFTP/FTP software upgrade
- Remote configuration/ reset
- LED indication for system status
- Support Fix IP and DHCP

Audio feature

- Codec -- G.711 a/μlaw, G.723.1 (6.3kbps), G.729, G.729A
- G.168/165-compliant adaptive echo cancellation
- Dynamic Jitter Buffer
- Completed voice band signaling support
- Provide In-band or RFC2833 DTMF generation/detection
- Provide call progress tone

Management Features:

Three easy ways for system configuration

- Console port: RS-232C port
- TELNET
- HTTP Brower (e.g. Internet Explorer)

Certification

- UL, CE, FCC

FXO Features

- 2-wire loop start
- Support auto-attendant (Tone or voice greeting)

- PSTN polarity reversal detection
- Disconnect tone detection
- Asking ping function with the incoming calls from PSTN side
- Record and analyze the Tone from PSTN side

Environmental

- Operation temp:0°C to 40°C
- Humidity: 10% to 90% (Non-condensing)

1.3 Accessories and equipment

- The voice gateway in 2/4 FXO ports models and only one RJ-45 connector (WAN).
- The AC adapter.
- The CD of user manual.
- The connection cable in RS-232 interface.

1.4 Appearance

Front panel: The LED lights provide related system messages of the gateway.

2FXOA

4FXOA

Power: Light on means Gateway is power on, and vice versa.

LINE: Light on means the line is in use (off-hook), and vice versa.

Status:

1. LED light on means Gateway has successfully registered to Proxy when it is in the Proxy Mode.
2. LED flash means Gateway is not registered to the Proxy when it is in the Proxy Mode.
3. Or when Gateway is in downloading mode, LED should be flash as well.
4. LED light off means Gateway is in Peer-to-Peer Mode.

Ready:

1. Light on and slow flash means Gateway is in normal mode.
2. Light on and fast flash means Gateway is in downloading mode.

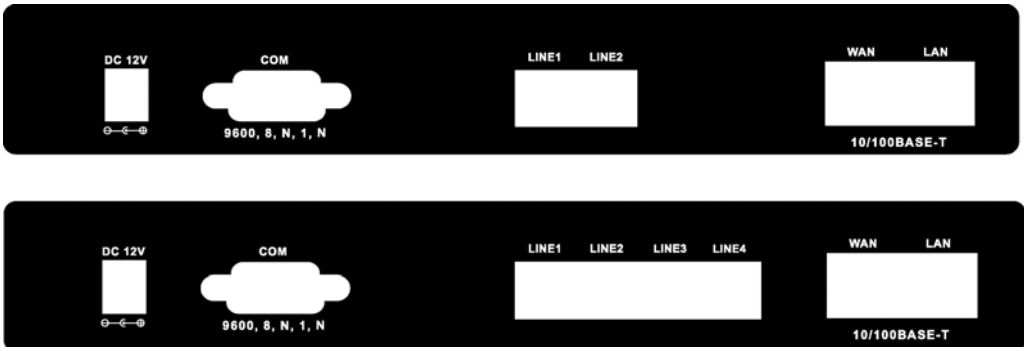
WAN/LAN

10/100: Network Rate to be 10 Base-T or 100 Base-T.

Link: Light on means the Gateway was connected with the network connector.

Act: LED should be light on and in flash display when data is transmitting.

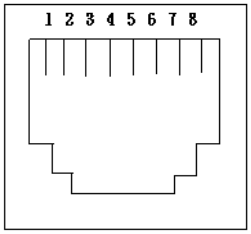
Back panel:



1. **Ethernet Port**

   LAN/WAN: 10/100 Base-T; RJ-45 socket, complied with ETHERNET 10/100base-T.

   The pin-out is as following:



PIN 1, 2: Transmit
PIN 3, 6: Receive

2. **LINE:**

   RJ-11 connector, FXO interface is for connecting the extension line of PABX or PSTN Line.

3. **COM:**

   RS232 console port (DB-9pin **male** connector)

   Note: use straightforward cable to connect to your computer.

PIN OUTS

Pin	Name	Dir	Description
2	RXD	←	Receive Data
3	TXD	→	Transmit Data
5	GND	—	System Ground

4. 12V DC:

Input AC 100V~120V; output DC12V.

2. How to Setup and connect basically

2.1 System Requirement

1. One PC (a) Pentium 100 or above, 64 RAM, Windows 98 or above.
   (b) Ethernet card or COM port
2. One standard straightforward RS-232 cable (female connector to Gateway side).
3. PBX extension Lines or PSTN Lines.
4. Software tools (a) Hyper Terminal, TELNET, WEB Browser.

2.2 IP Environment Setting

User must prepare a valid IP address, complied with IP Network, for Gateway's proper operation.

For testing the validation of chosen IP address, using the same IP configuration in other PC or Notebook, and then try to connect to Public Internet (go to well-known website, receive Internet mail, or ping a specific public IP address). If it works, use the same IP address and network configuration for Gateway.
Please follow up the step for the configuration of your computer or notebook.

2.2.1 For Windows 2000/NT

Please make sure that the network interface of your computer is working fine and the cross over line (RJ-45) is connecting with the computer correctly or you could use a hub to connect with your computer and this gateway. Turn on your computer and configure the network parameter as follow:

1 Go to the **start** menu and enter the **setting** area. Click **control panel**.

2 Enter the network configuration.

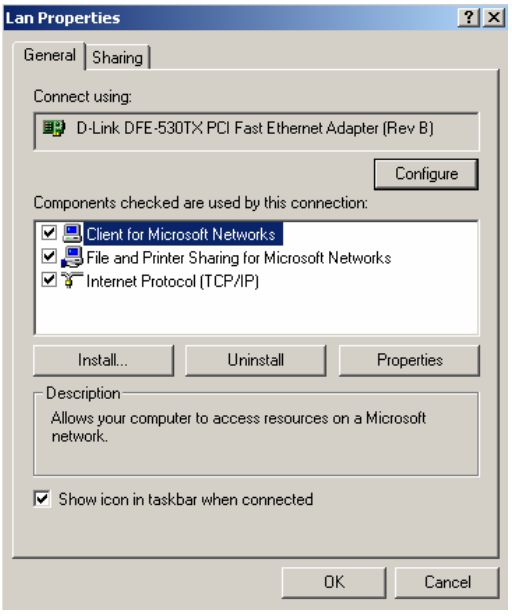


Figure 2.1: Network Configuration

3 Select the **Property** of the LAN card.

4 Setup the ip address, subnet mask and default gateway as below:

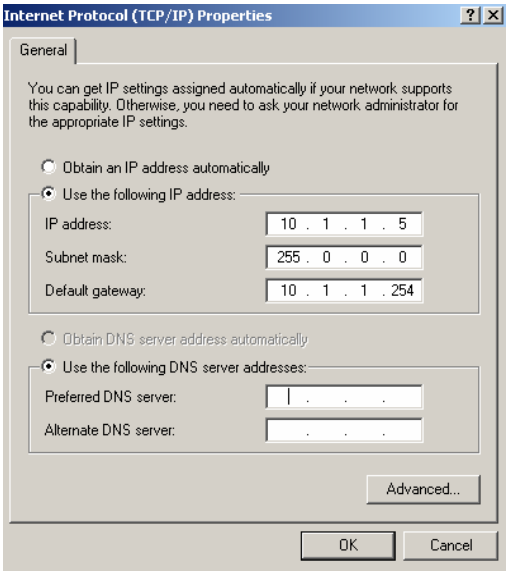

Internet Protocol (TCP/IP) Properties
General
You can get IP settings assigned automatically if your network supports this capability. Otherwise, you need to ask your network administrator for the appropriate IP settings.
Obtain an IP address automatically
Use the following IP address:
IP address: 10 . 1 . 1 . 5
Subnet mask: 255 . 0 . 0 . 0
Default gateway: 10 . 1 . 1 . 254
Obtain DNS server address automatically
Use the following DNS server addresses:
Preferred DNS server:
Alternate DNS server:
Advanced...
OK
Cancel

Figure 2.2: Configure the network

5 Click OK after you finished the network setup.

The default ip address, netmask and default gateway address of the gateway is 10.1.1.3, 255.0.0.0, 10.1.1.254.

2.3 Network configurations in your gateway

1 Key in the ip address of the gateway (http://10.1.1.3) with the browser

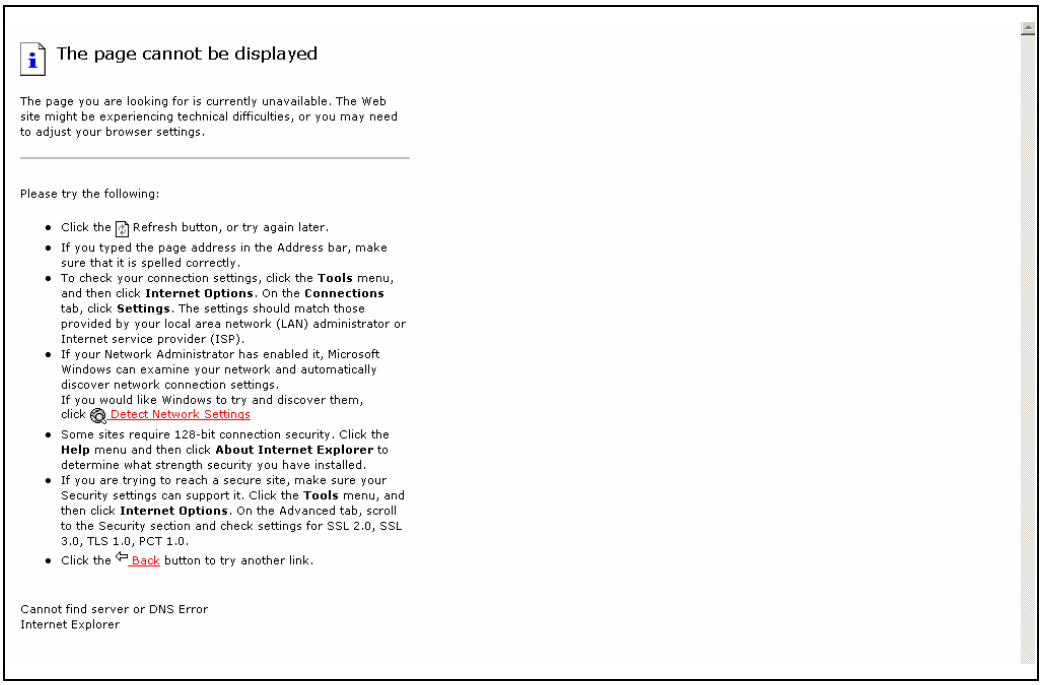

The page cannot be displayed
The page you are looking for is currently unavailable. The Web site might be experiencing technical difficulties, or you may need to adjust your browser settings.
Please try the following:
Click the Refresh button, or try again later.
If you typed the page address in the Address bar, make sure that it is spelled correctly.
To check your connection settings, click the Tools menu, and then click Internet Options. On the Connections tab, click Settings. The settings should match those provided by your local area network (LAN) administrator or Internet service provider (ISP).
If your Network Administrator has enabled it, Microsoft Windows can examine your network and automatically discover network connection settings.
If you would like Windows to try and discover them, click Detect Network Settings
Some sites require 128-bit connection security. Click the Help menu and then click About Internet Explorer to determine what strength security you have installed.
If you are trying to reach a secure site, make sure your Security settings can support it. Click the Tools menu, and then click Internet Options. On the Advanced tab, scroll to the Security section and check settings for SSL 2.0, SSL 3.0, TLS 1.0, PCT 1.0.
Click the Back button to try another link.
Cannot find server or DNS Error
Internet Explorer

Figure 2.3: WEB Browser

2 After key in the ip address, you have to enter the user name and password to enter the WEB configuration. (Username: root ; No password)

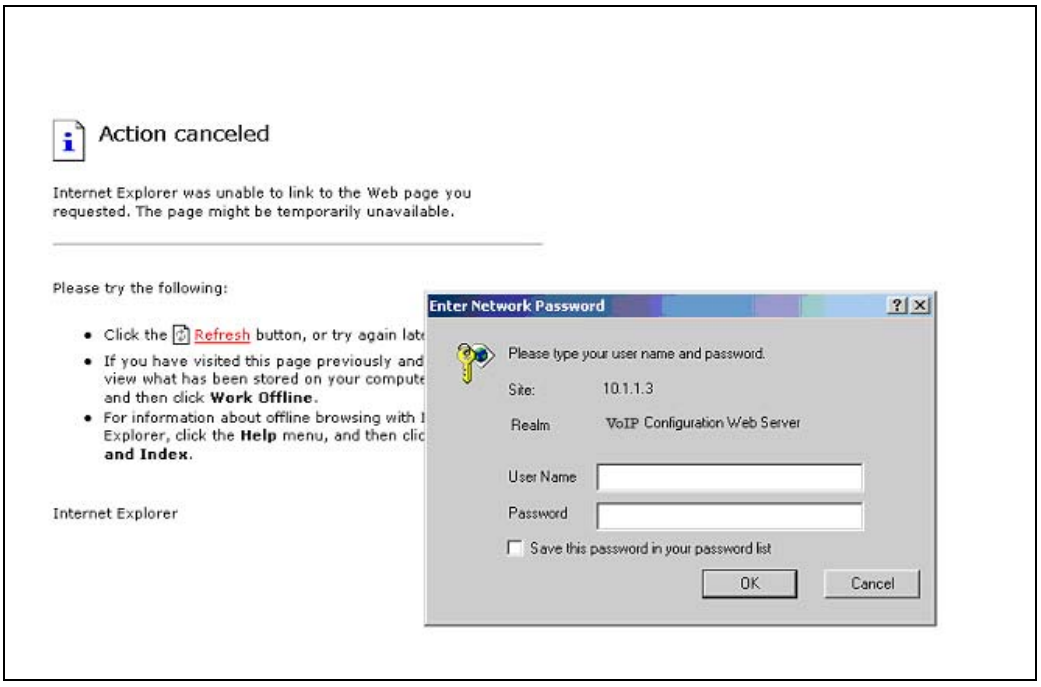

Action canceled
Internet Explorer was unable to link to the Web page you requested. The page might be temporarily unavailable.
Please try the following:
Click the Refresh button, or try again late
If you have visited this page previously and view what has been stored on your compute and then click Work Offline.
For information about offline browsing with I Explorer, click the Help menu, and then clic and Index.
Internet Explorer
Enter Network Password
Please type your user name and password.
Site: 10.1.1.3
Realm VoIP Configuration Web Server
User Name
Password
Save this password in your password list
OK
Cancel

Figure 2.4: Login the username and password

3 You will enter the main page of the configuration after key in the login name and password correctly:

Figure 2.5: The main WEB configuration

4 Press the ***Network Interface*** to configure the networking of your gateway:

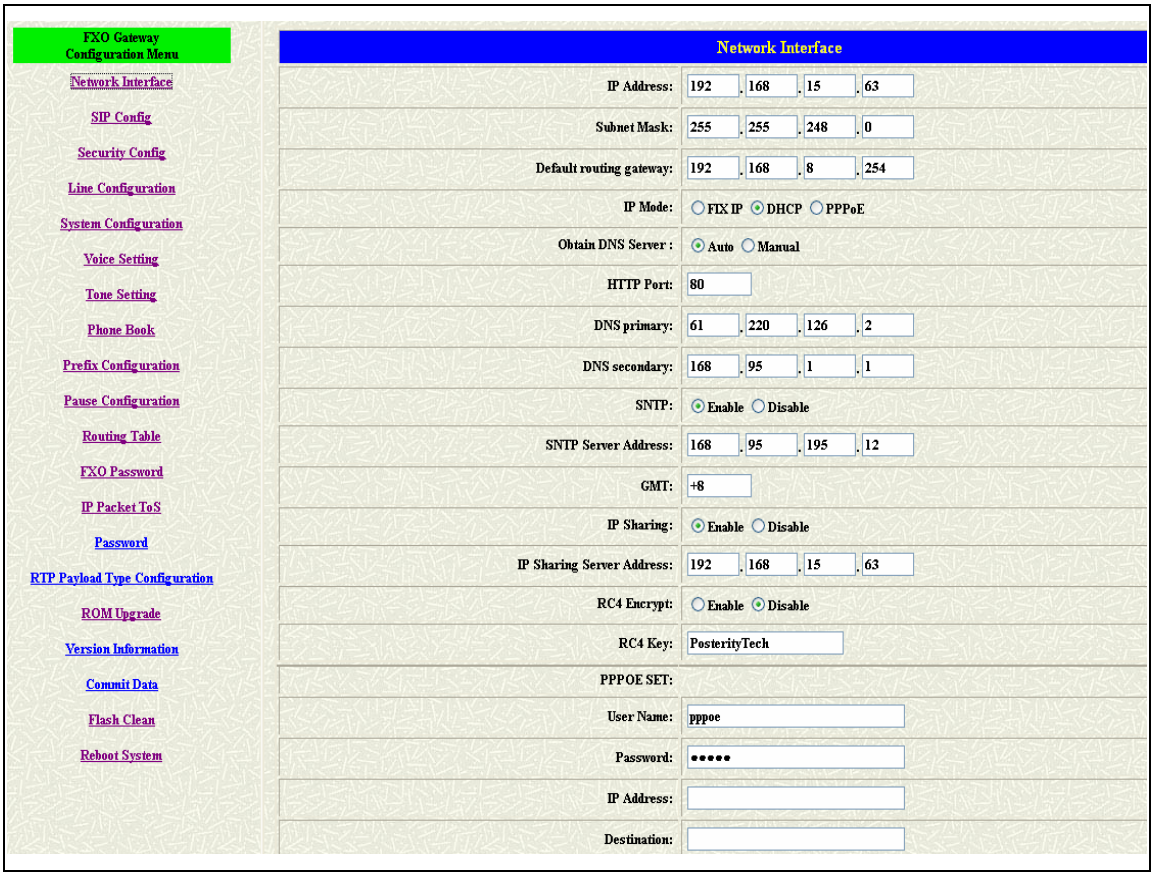


Figure 2.6: The Network Interface

2.3.1 Static ip address

1 Please get the correct ip address, netmask and default gateway address from your ISP first. Press the OK button if you finished.

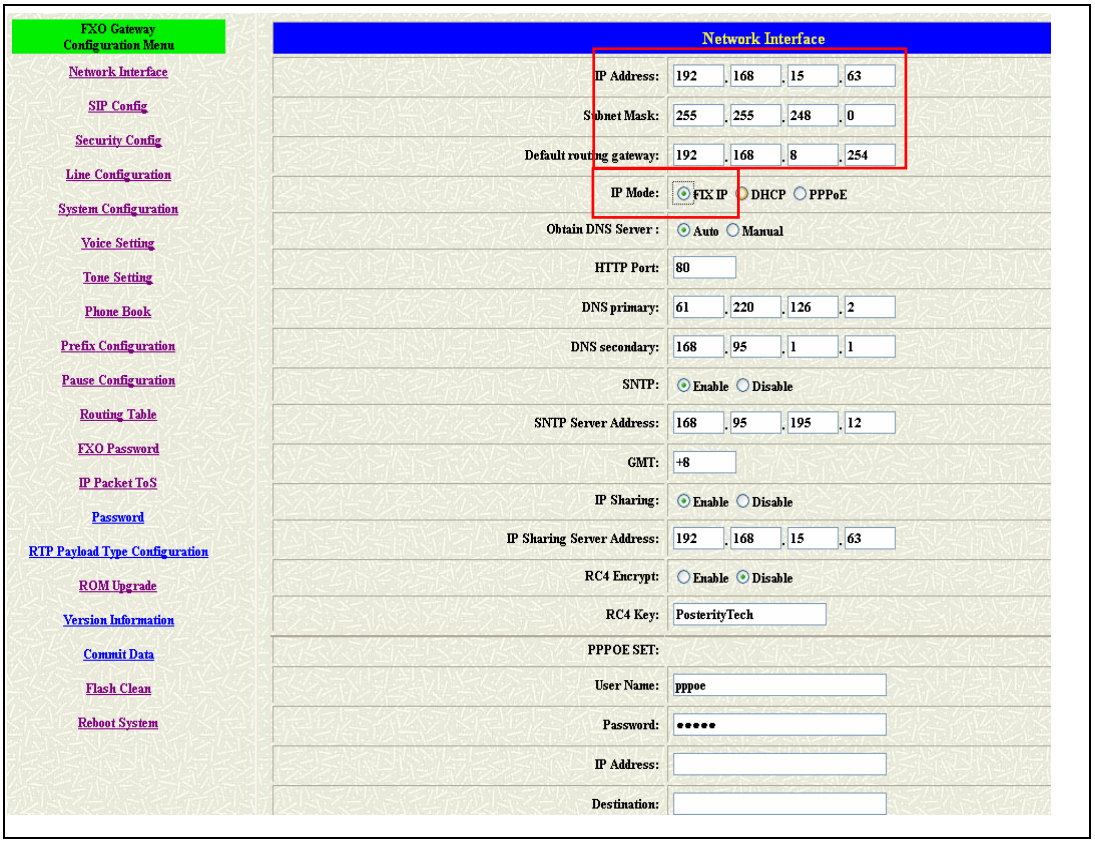


Figure 2.7: Configure the static ip address

2 Press the commit if you finish the configuration.

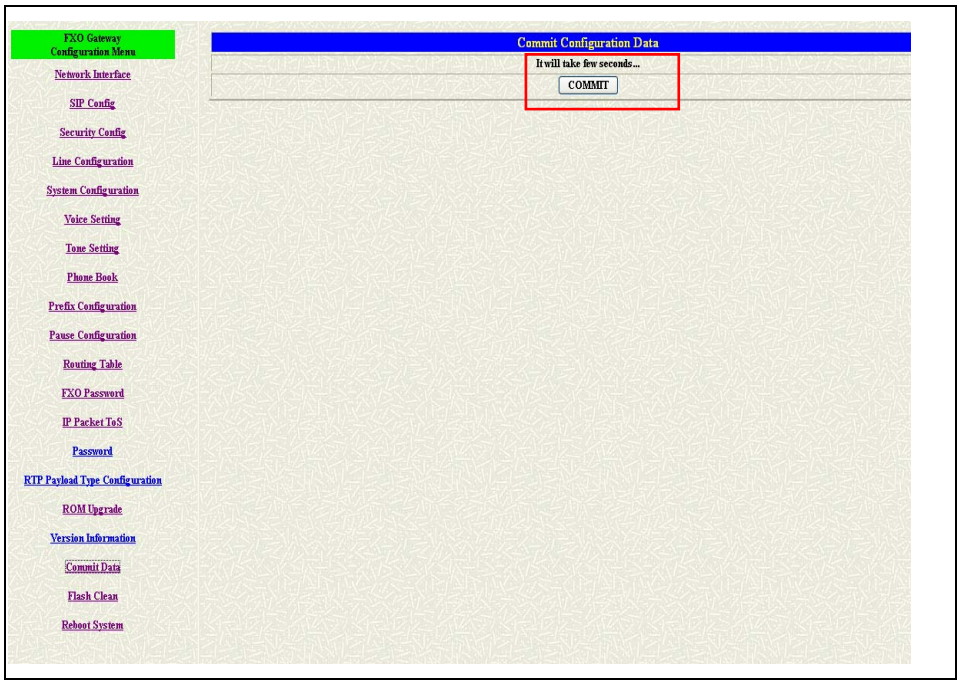


Figure 2.8: Commit the data

3 Press the reboot if you want the configuration executed.

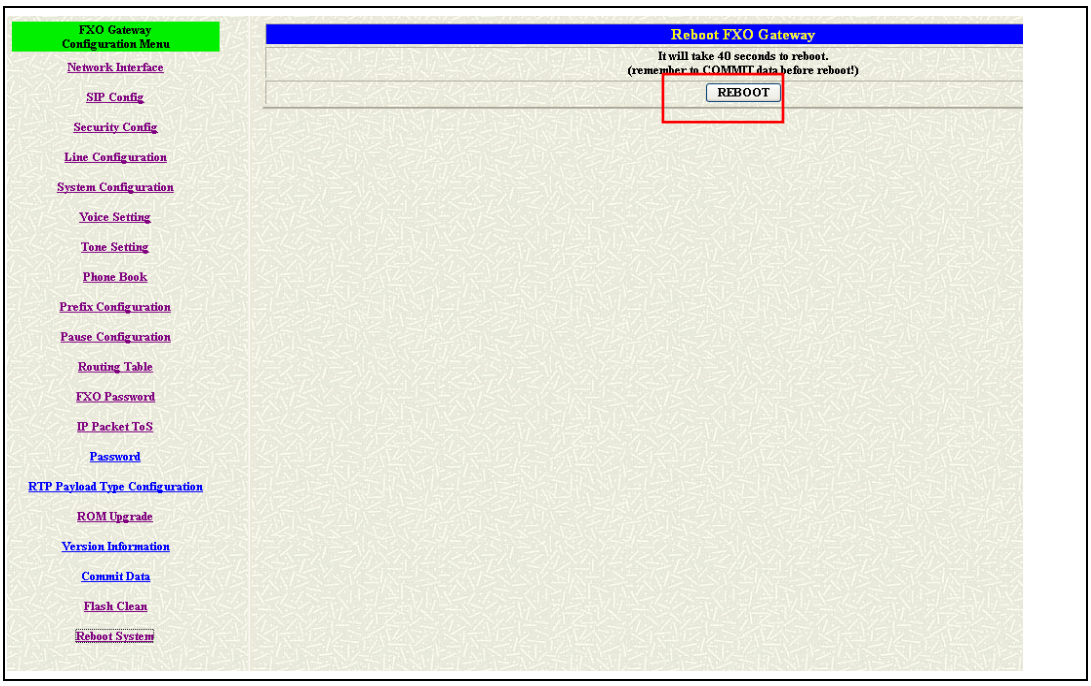


Figure 2.9: Reboot the system

2.3.2 DHCP mode

1 Enable the DHCP if you are using the cable modem or DHCP server.

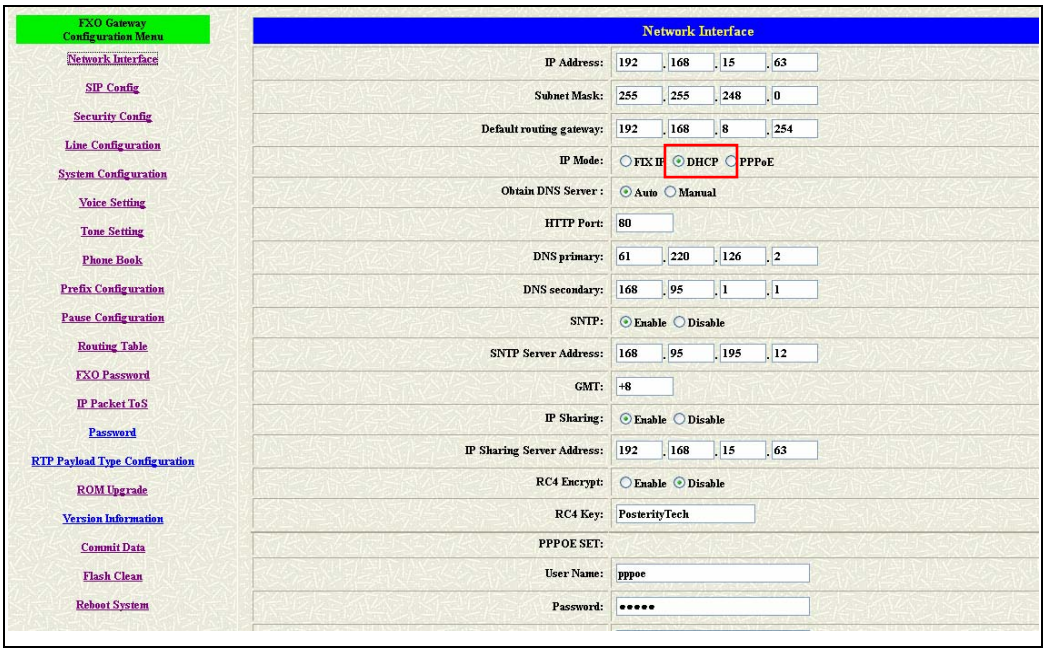


Figure 2.10: Enable the DHCP function

2 Please commit the data and reboot the machine after you enable the DHCP

function.

2.3.3 PPPoE mode

1 Enable the PPPoE and press the info about PPPoE configuration in the correct table.

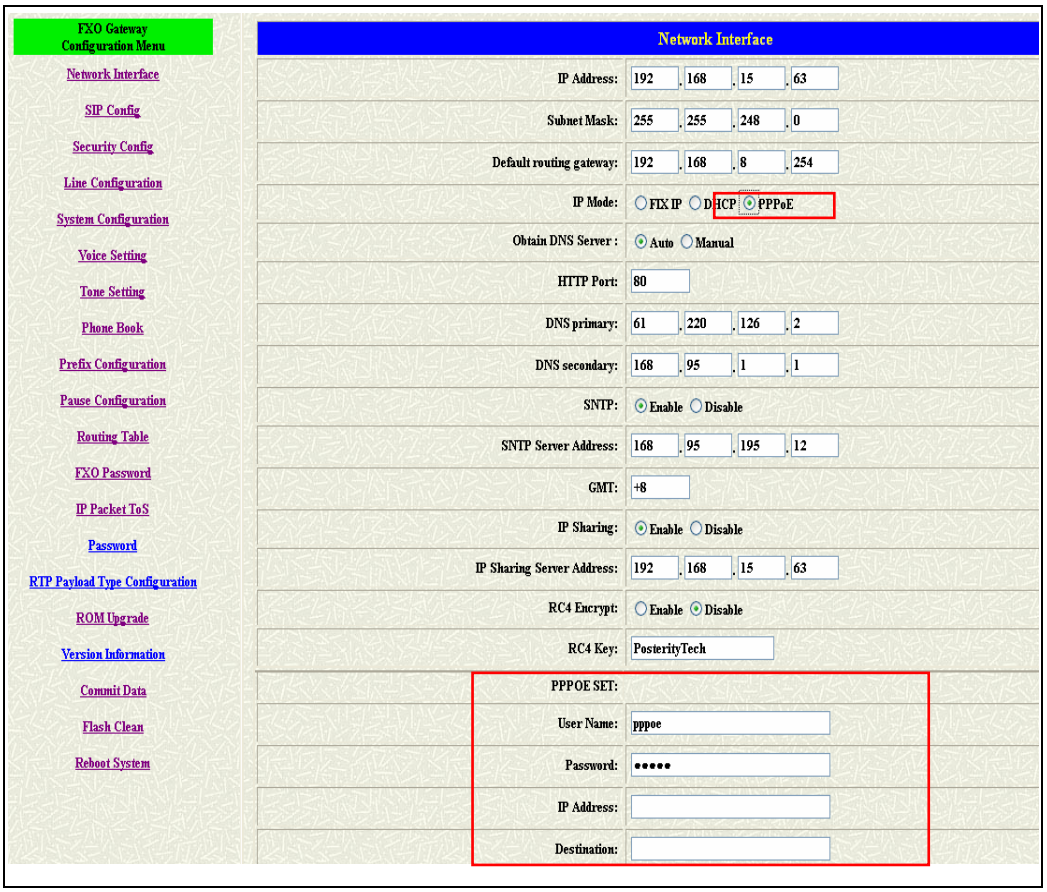


Figure 2.11: Switch to PPPoE function

2 Please commit the data and reboot the machine after you finished the configuration of PPPoE.

2.4 Making a VoIP Call

There are two modes that you could configure the gateway for making VoIP calls. One is the Peer-to-Peer mode, another is Proxy mode. The

configurations and functions are different. Please make sure about the mode you want and follow up the step to configure your gateway.

2.4.1 Configure the gateway into the Peer-to-Peer mode

1 Enter the ***SIP Configuration*** table and change the mode to Peer-to-Peer. Define the port numbers whatever you like. Press the "OK" button if the configuration is all finished. (see figure 2.13)

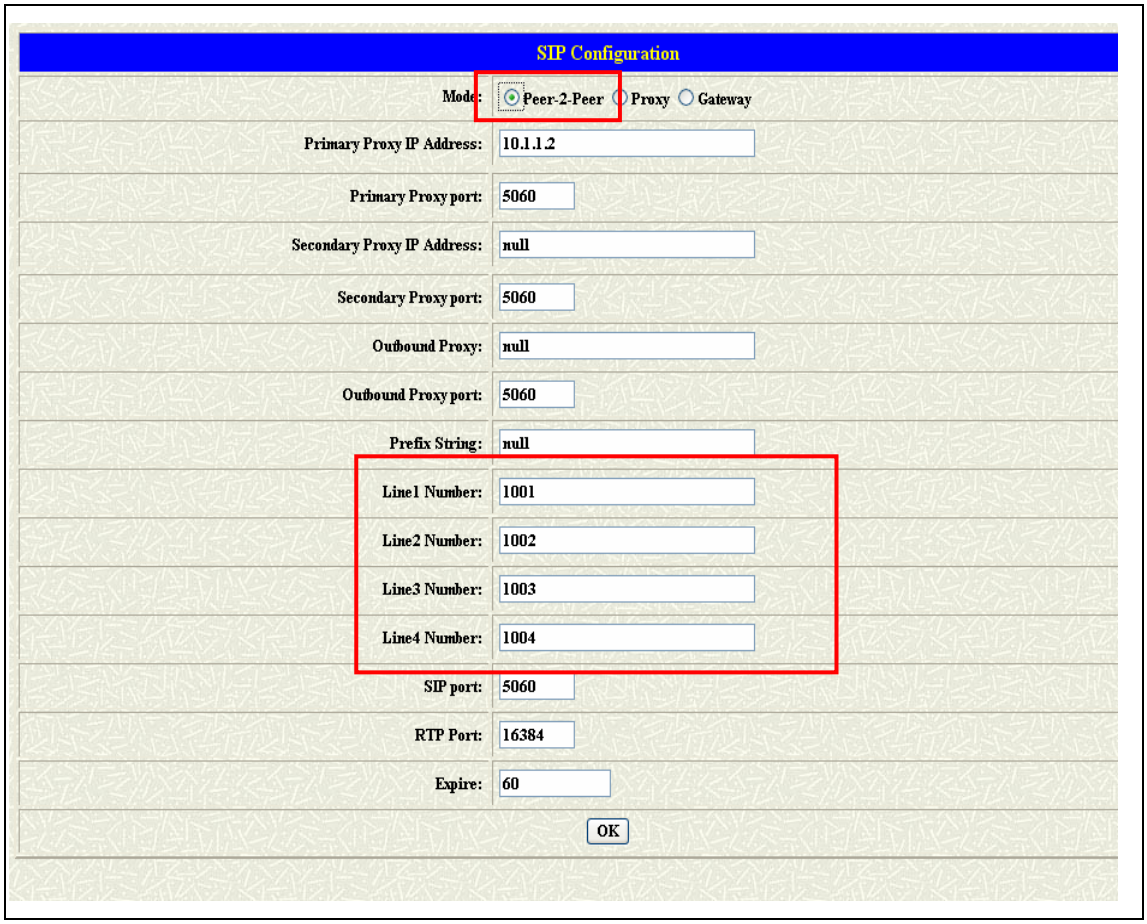


Figure 2.13: Configure the Peer-to-Peer mode

2 Enter the ***Phone Book configuration*** table and configure the name, ip address and phone number of the destination. (see figure 2.14)

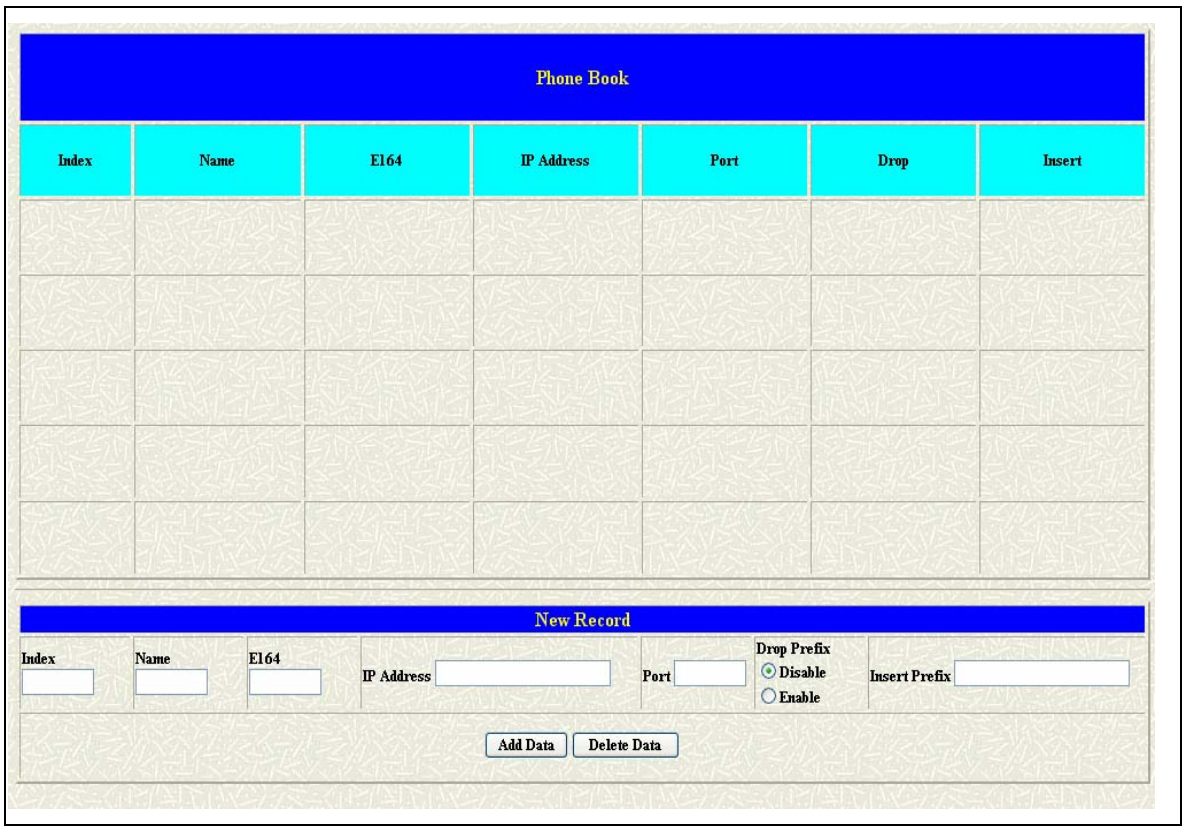


Figure 2.14: Phone Book

【Example】

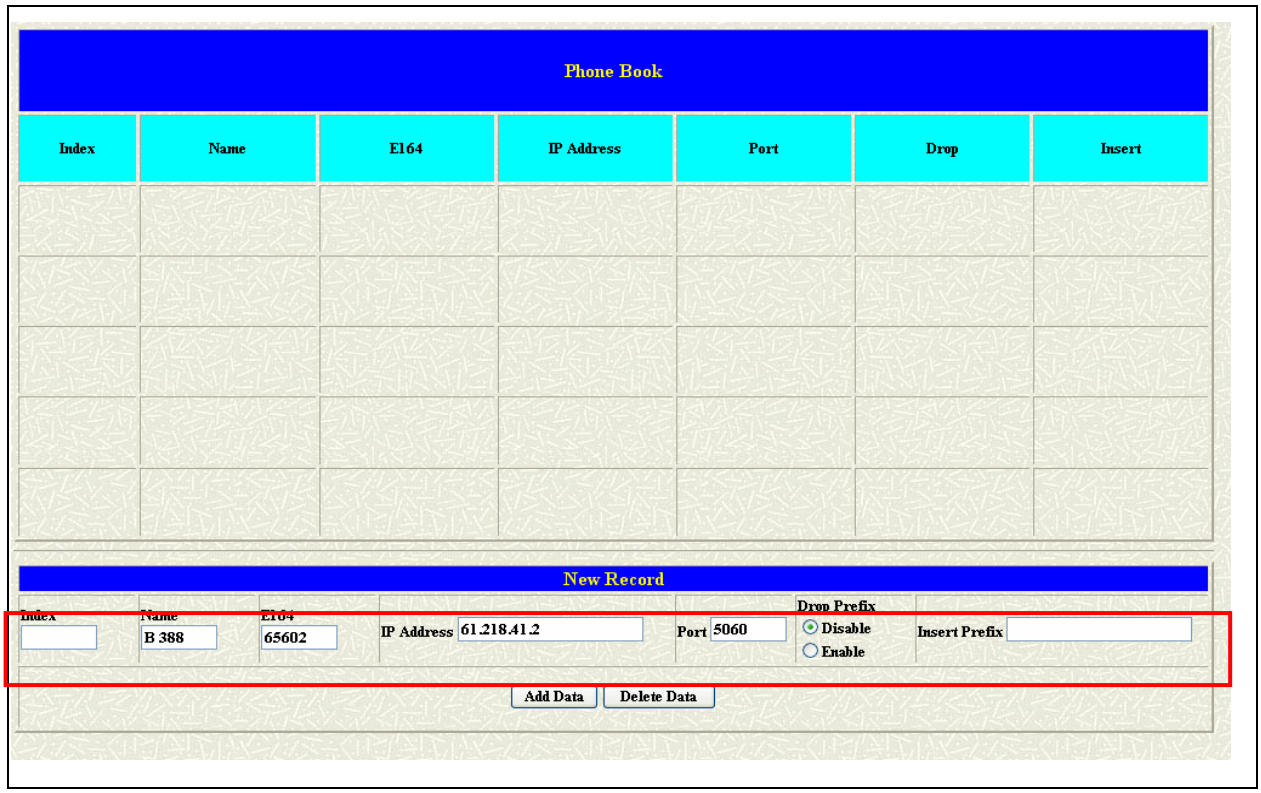


Figure 2.15: Phone Book

This is the first record of Phone Book. So the index is **1**
The name of the destination: **B 388**
The E164 number (phone number) of the destination: **65602**
The ip address of the destination: **61.218.41.2**
The port of the destination: **5060**

4 Press the “Add Data” button when you finished, and the new table will display on the first index if you press the Phone Book configuration button.

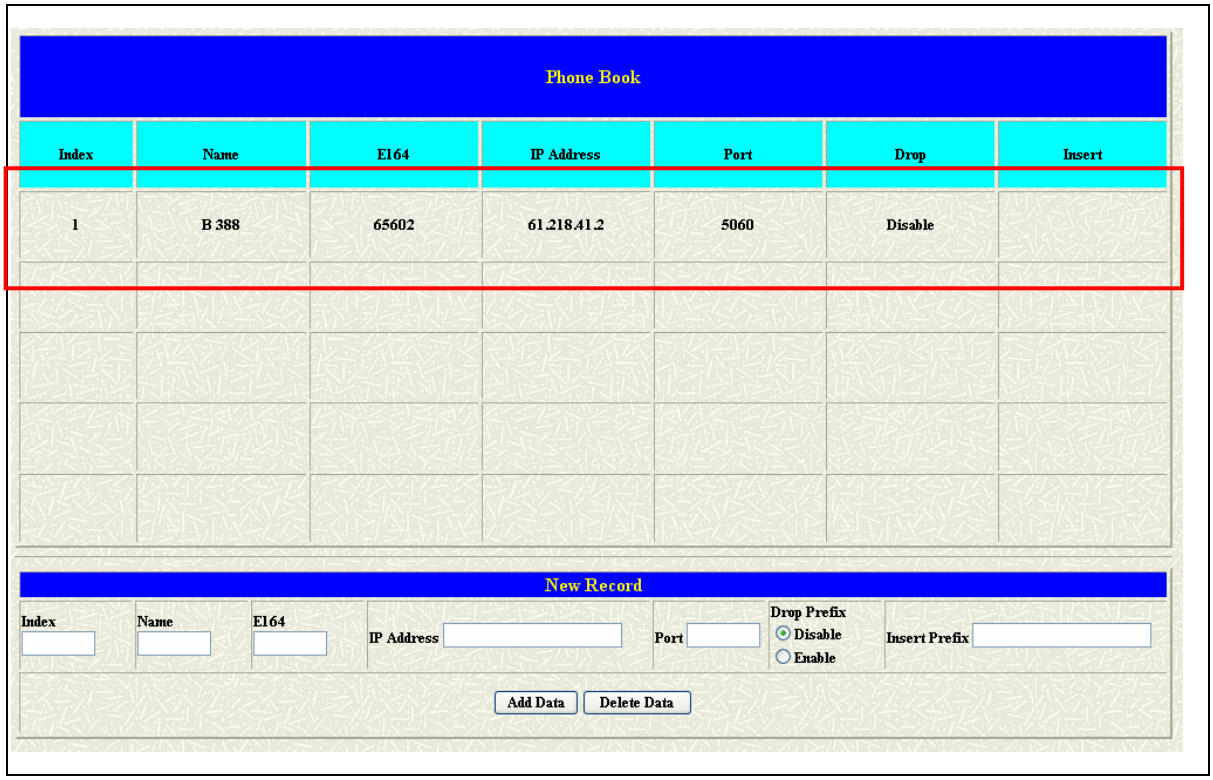


Figure 2.16: To show the Phone Book record

5 Please Commit it and Reboot the system if the configuration is finished.

Phone Book is only for the Peer-to-Peer mode and could support forty records.

【The application in the drop and insert function】

Input (E164)	Drop	Insert	Output
100	Disable	X	100
200	Disable	0	0200
300	Enable	X	X
400	Enable	500	500

※ X – Do not enter any numbers

Please make sure about that the FXO port was connected with the PSTN line or the extension line of the PABX before you start to make a voip call.

2.4.2 Configure the gateway into the Proxy mode

1 Enter the ***SIP Configuration*** table and change the mode from Peer-to-Peer to Proxy mode. To change the Proxy information from your service provider (Ex: The Proxy IP, Outbound Proxy IP and Registered number).

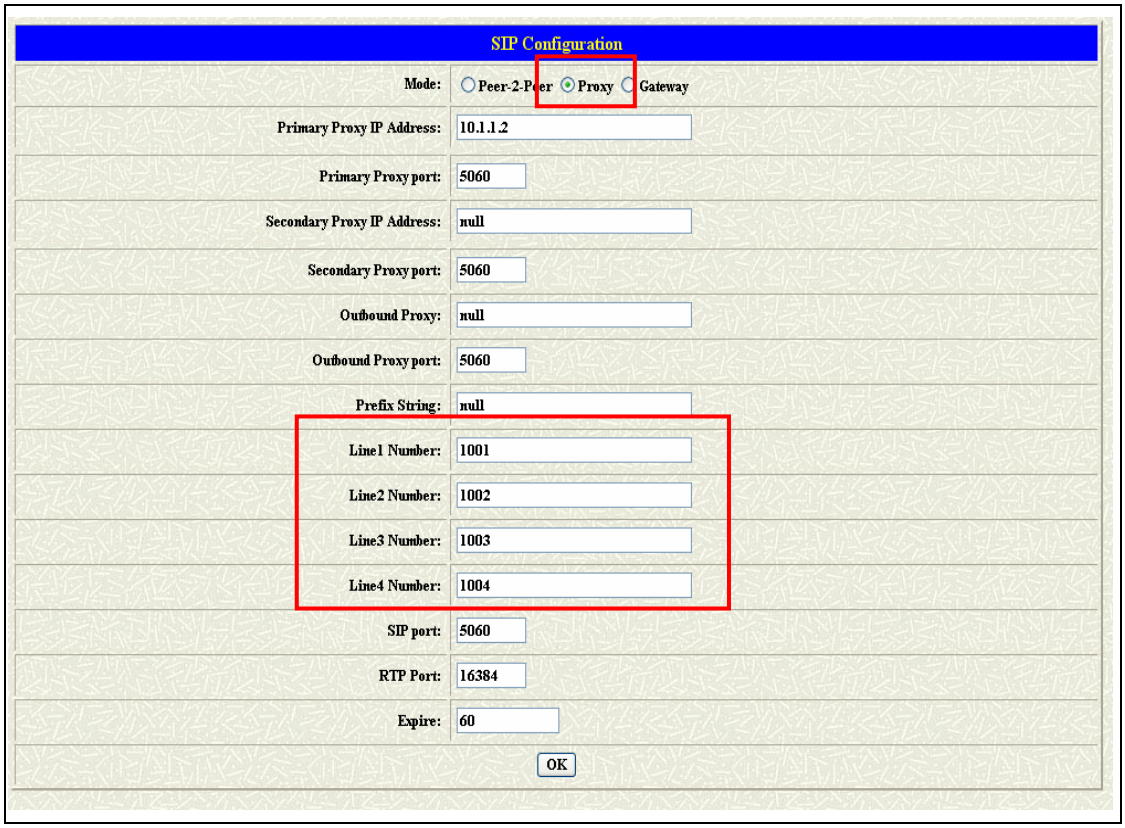


Figure 2.17: Configure the Proxy info

2 Press the OK button that is on the bottom of this page to save the configuration.

3 Switch to the ***Security Config*** page and put the user account and password in the correct table. Please get this info from your ITSP. Press the OK button if the configuration is finished. (see figure 2.18)

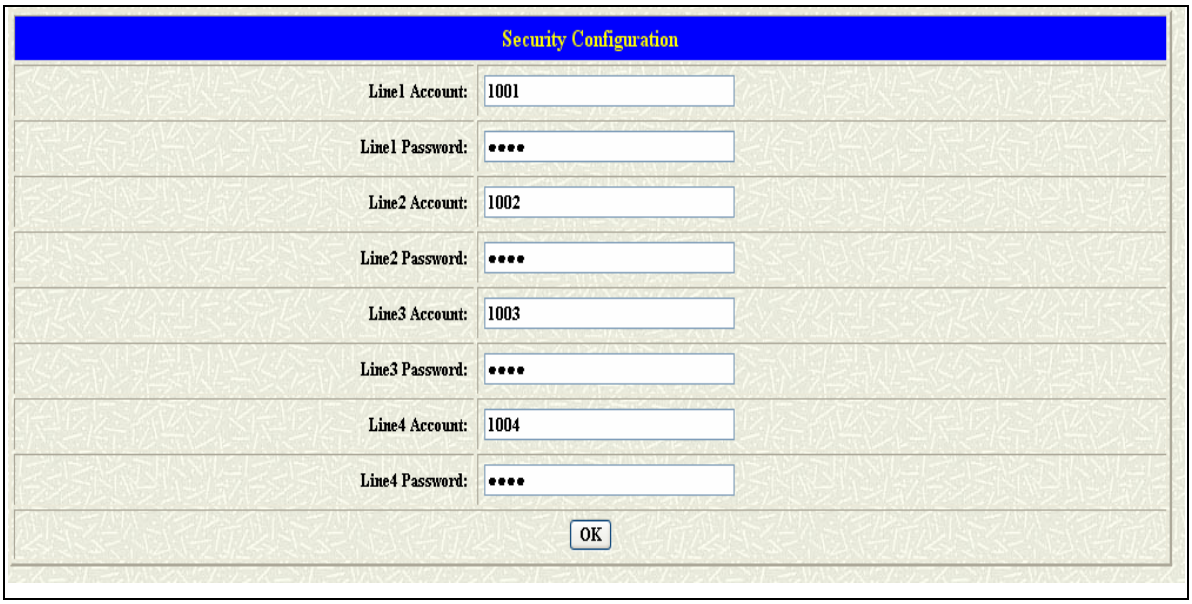


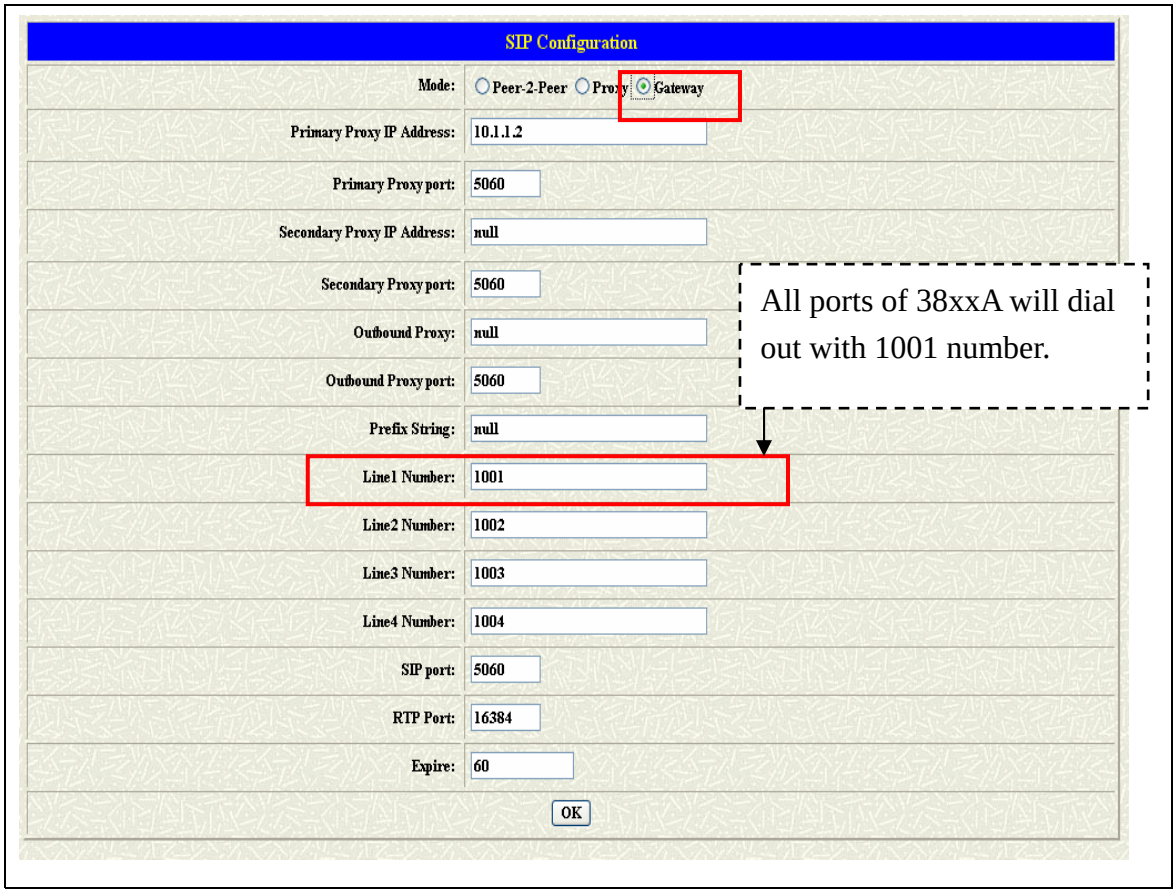
Figure 2.18: Configure the Security info

4 Press the Commit Data and Reboot System buttons when you finished the configuration.

2.4.3 Configure the gateway into the Gateway mode

1 Enter the SIP Configuration table and change the mode from Proxy to Gateway mode. To change the Proxy information from your service provider (Ex: The Proxy IP, Outbound Proxy IP and Registered number).

*** Gateway**: All ports of 38XXA will use one Line number on them (PX limits the account just can be register by one), user can pick up this option and just input Line1 number &account & password, the all ports will use this Line1 number on the same time.



2 Press the OK button that is on the bottom of this page to save the configuration.

3 Switch to the ***Security Config*** page and put the user account and password in the correct table. (Just input Line1 account & password) Please get this info from your ITSP. Press the OK button if the configuration is finished.

4 Press the Commit Data and Reboot System buttons when you finished the configuration.

3. Advance Configurations

There are too many advanced commands for the advanced users. The following chapters are based on the application layer. Please get the info what you need. If you need the command, please watching the chapter of Command Line Interface.

3.1 Network Interface

Users have to configure the Network configurations in this page. This gateway will be work while it is connecting with the internet network. Please get more info from the following descriptions. (see figure 3.1)

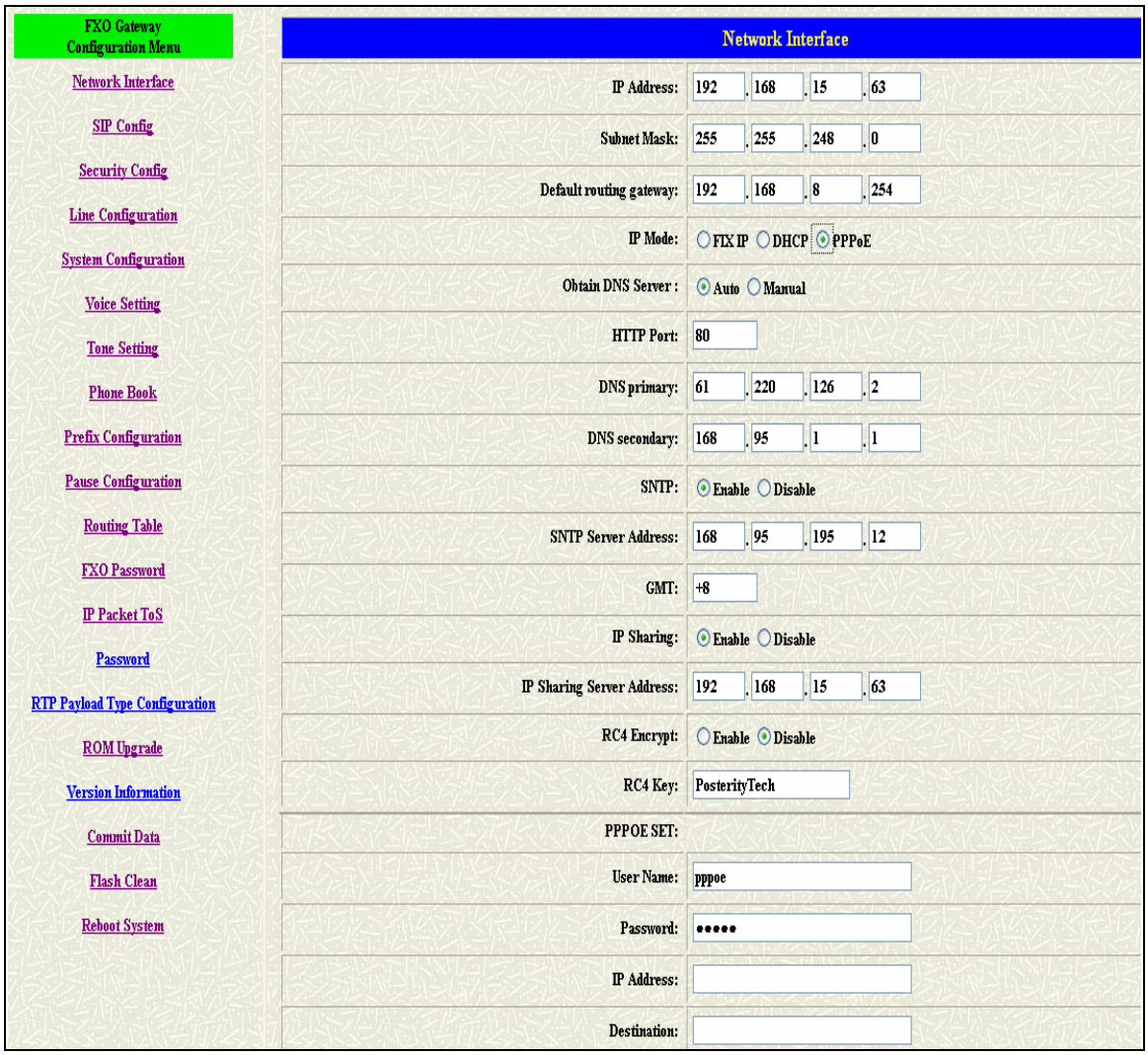


Figure 3.1: Network Interface

- IP Address – Define the ip address for your networking if it is the fixed ip. Please get this info from your ISP.
- Subnet Mask – Define the mask address for your networking. Please get this info from your ISP.
- Default Routing Gateway – Define the default gateway for your networking. Please get this info from your ISP.
- IP Mode – To configure the fixed or dynamic ip address for this unit. Please configure to PPPoE if the ADSL is using the PPPoE type.
- Obtain DNS server – User can input manual DNS address, but the item must be choice on “Manual” first.
- HTTP Port – To configure the HTTP port for access this unit from the remote side.
- DNS primary – To manually configure the first ip address for the DNS server.
- DNS secondary – To manually configure the second ip address for the DNS server.
- SNTP – Enable the SNTP server registering function if user wants to get the correct time from the Command Line Interface.
- SNTP Server Address – Enter the correct ip address of the SNTP server or get the incorrect time from the Command Line Interface.
- GMT – Configuring the time area for the time display in the Command Line Interface.
- IP Sharing – When behind NAT use. In order to through NAT device. After enable it, user must input Public IP on “IP Sharing sever address” item.
- IP Sharing sever address – Input Public IP of NAT device.
- RC4 Encrypt – Encryption function. For specially sever.
- RC4 key – Encryption function. For specially sever.
- PPPoE User Name – To configure the user name for the PPPoE connection.
- PPPoE Password – To configure the password for the PPPoE connection.
- PPPoE IP Address – In the PPPoE mode, this table will show the ip address that this unit gets from the ISP.
- PPPoE Destination – In the PPPoE mode, this table will show the default gateway address that this unit gets from the ISP.
- PPPoE DNS primary – In the PPPoE mode, this table will show the DNS ip address that this unit gets from the ISP.
- After Remote Host Disconnection – This unit will reboot and re-connect to

the ISP

- PPPoE Echo Request – the function is keep up IP connect with PPPoE sever, when network fail it can re-discover IP from PPPoE server.

3.2 SIP Information

This WEB page will help user to configure the information about the dial mode, P2P/ PX/ Gateway information and some port settings for the communication. Please get more info about this configuration from the below detail descriptions. (see figure 3.5)

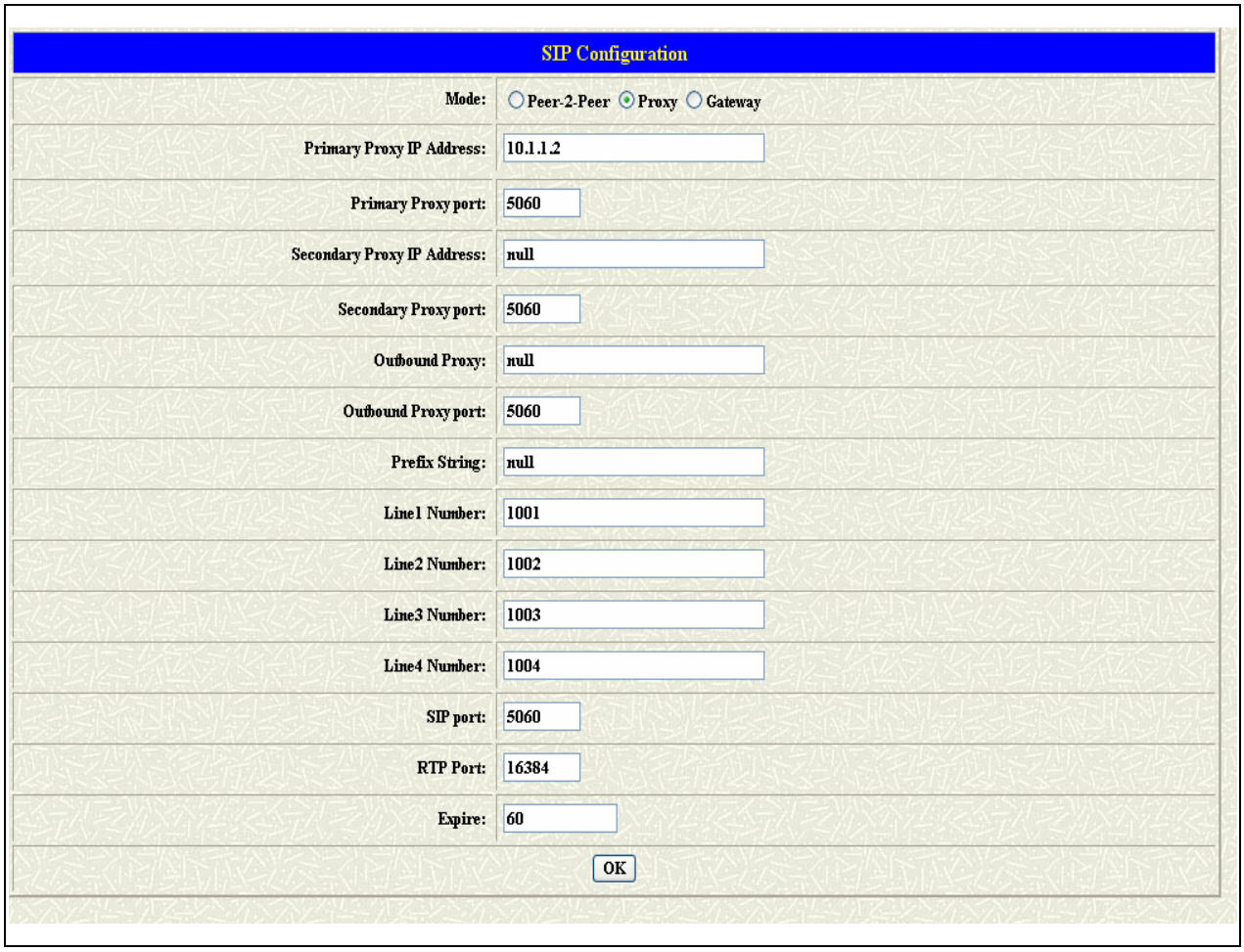


Figure 3.2: SIP Information

- Mode – Pick up the calling mode for this gateway.
  - Peer-2-Peer: It only supports the peer-to-peer mode and users have to define the phone book for this mode.
  - Proxy: Users have to register on the Proxy if users picked up this

option.

- Gateway: The all ports of 38XXA, if user wants to use one Line number on them (PX limits the account just can be register by one), user can pick up this option and just input Line1 number &account & password, the all ports will use this line1 number on the same time.

- Primary Proxy IP Address – Enter the proxy ip if users pick up the proxy mode or Gateway mode.
- Primary Proxy port – Set Proxy port for 380x to send message, default value is 5060, if there is no special request of Proxy server, please don't change this value.
- Secondary Proxy IP Address – Set secondary Proxy IP Address or URL address (Domain Name Server must be configured. Please refer to Network Interface). When 380x fail to register to primary Proxy, it will try to register to secondary Proxy, when it fails again, it will retry to register to Primary Proxy.
- Secondary Proxy port – Set the secondary proxy port for every SIP message, default value is 5060.
- Outbound Proxy – This version could support the outbound proxy, users could define the ip address or domain name in this table.
- Outbound Proxy port –Set outbound Proxy port for 380x to send message, default value is 5060, if there is no special request of Proxy server, please don't change this value.
- Prefix String – Users could define this if the registration name was a phonetic alphabet not the numbers.
- Line 1 Number – The phone number of the Tel 1.
- Line 2 Number – The phone number of the Line 1.
- Line 3 Number – The phone number of the Tel 2.
- Line 4 Number – The phone number of the Line 2.
- SIP port – Users could change the sip port of this unit for the registration.
- RTP port – Users could change the beginning RTP ports in this table.
- Expire – Users could change the expire time for the register message sending.

3.3 Security Config

Some proxy will include the security policy. The endpoint may need the user account and password for the registration. If these are necessary, users could

put the correct account and password in the correct table. (see figure 3.3)

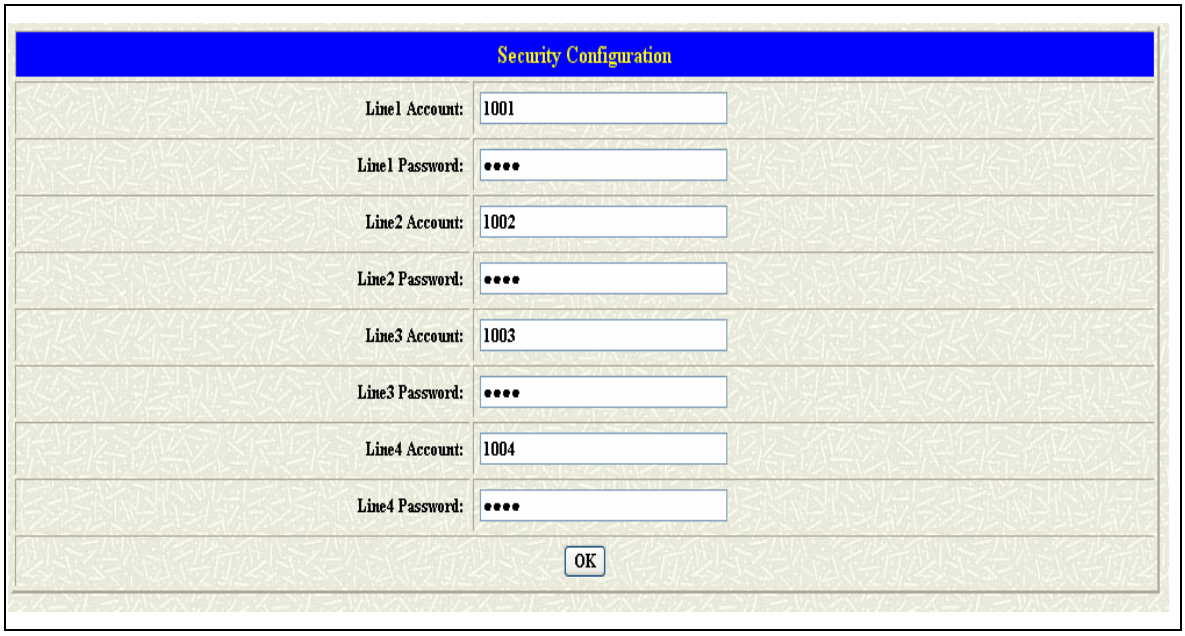


Figure 3.3: Security Configuration

- Line 1 Account – The user name for the line 1 account.
- Line 1 Password – The password for the line 1 account.
- Line 2 Account – The user name for the line 2 account.
- Line 2 Password – The password for the line 2 account.
- Line 3 Account – The user name for the line 3 account.
- Line 3 Password – The password for the line 3 account.
- Line 4 Account – The user name for the line 4 account.
- Line 4 Password – The password for the line 4 account.

3.4 Line Configuration

The Line configuration will show the status of the registrations and the ports. It includes the hunt group, hotline, and no answer forward configuration. Press the Line configuration button to enter configuration table (see figure 3.4)

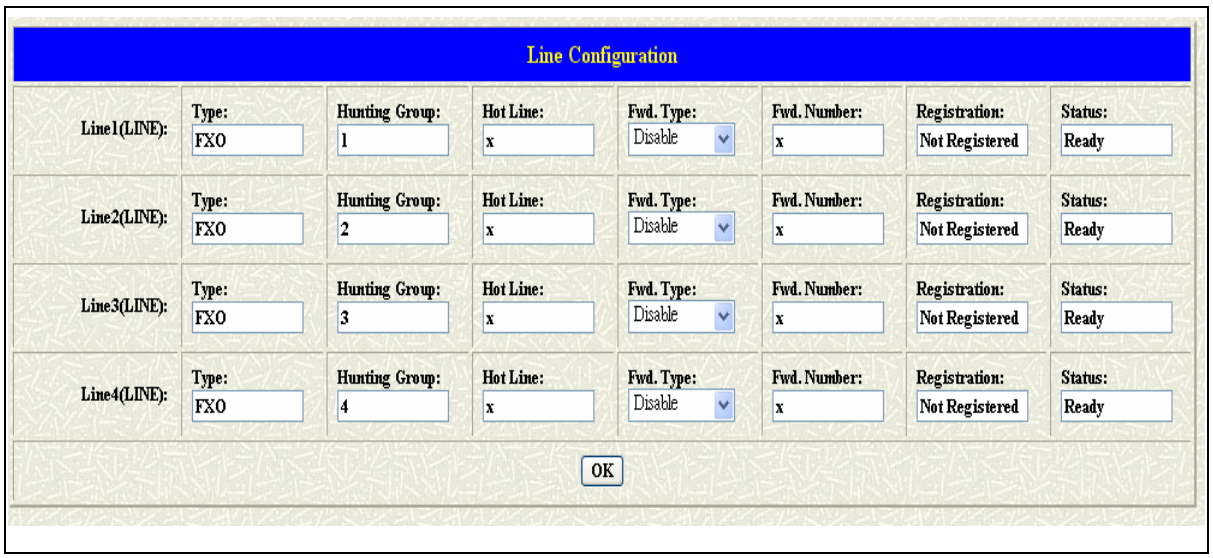


Figure 3.4: Line Information

- Type – Just show the interface for this port.
- Hunt Group –Define the group number of this port. When the port is busy, the call could be transferred to another port in the same group. Only the same type could be configured in the same group.
- Hotline –Enable or Disable the hotline mode. The hotline mode will be enabled if you enter the hotline number. The default setting is disabled.
- Registration –To show the gateway registered on the GK or not. Support the Proxy mode only.
- Status –To show the port is busy or ready.

3.5 System Configuration

There are some parameters in the system configurations, please get more detail as following. (see figure 3.5)

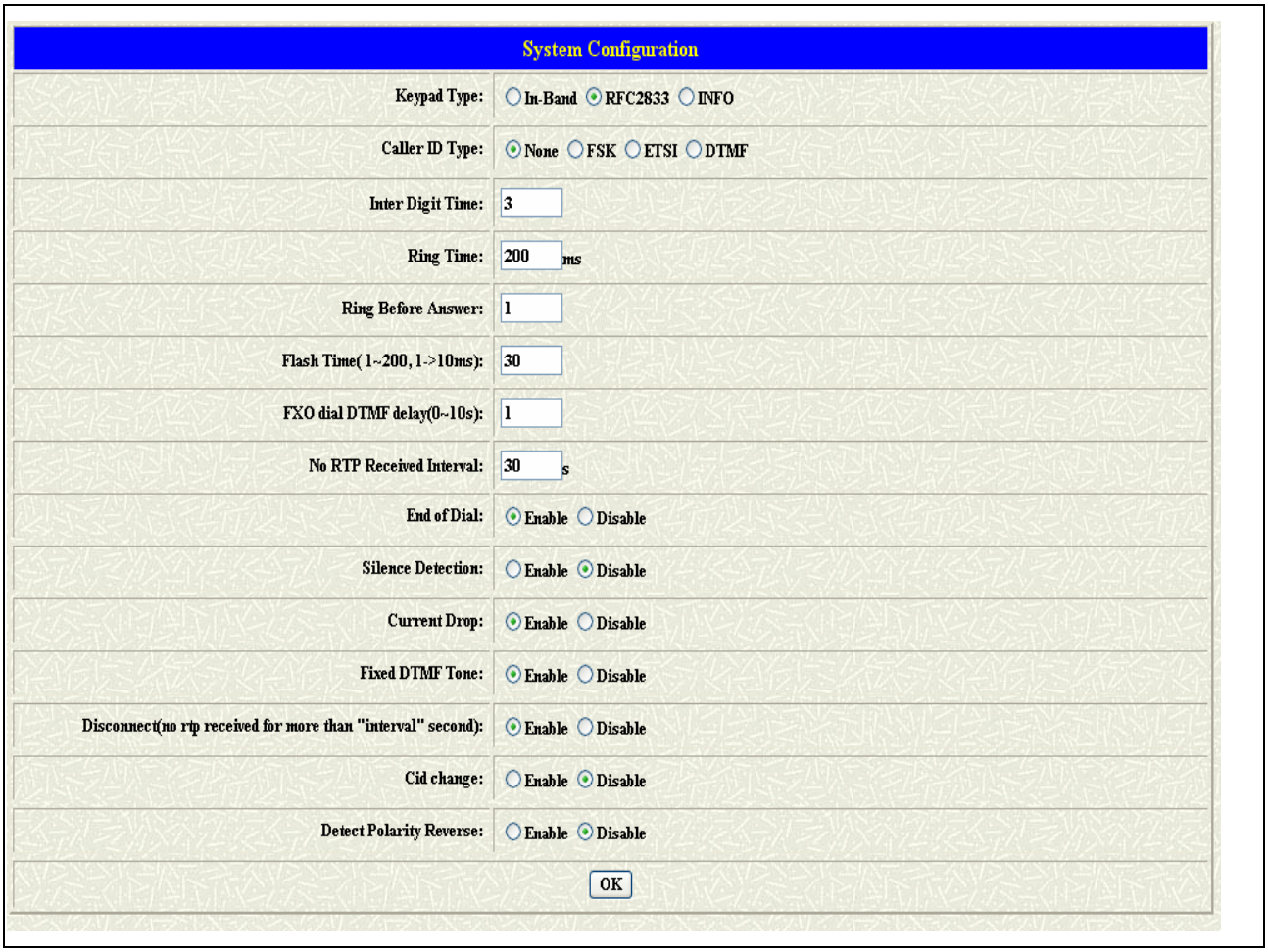

System Configuration
Keypad Type: In-Band RFC2833 INFO
Caller ID Type: None FSK ETSI DTMF
Inter Digit Time: 3
Ring Time: 200 ms
Ring Before Answer: 1
Flash Time(1~200, 1->10ms): 30
FXO dial DTMF delay(0~10s): 1
No RTP Received Interval: 30 s
End of Dial: Enable Disable
Silence Detection: Enable Disable
Current Drop: Enable Disable
Fixed DTMF Tone: Enable Disable
Disconnect(no rtp received for more than "interval" second): Enable Disable
Cid change: Enable Disable
Detect Polarity Reverse: Enable Disable
OK

Figure 3.5: System Configuration

- Keypad type – There are three types for the Keypad. On is the In-Band type, another is the RFC2833 type. User could define the keypad type for the dialing.
- Caller ID Type –There are three types for the Caller ID type detection from PSTN side. User must know which type PSTN side will send, then configure this item.
- Inter Digit Time – It's the time for the time out during the dialing numbers.
- Ring Time – FXO will detect the ring tone according this time.
- Ring Before Answer(FXO) – This will help the users to answer the calls from PSTN into this gateway quickly. The call will be connected by one time ring if users configure this for 1. (From 1 to 10)
- Flash Time – When FXO receives SIPINFO (16) message, it will produce "Flash Time" (300) to PSTN side. 1->10 ms.
- No RTP Received Interval – The parameter was used by "Disconnect (no rtp received for more than "interval" seconds" function.

- FXO dial DTMF delay – delay time to dial out to PSTN side.
- End of Dial – It will transfer the digit "#" if this function is disabling.
- Silence Detection – G711a/u & G729 VAD/CNG function.
- Current Drop –* "Current Drop" it is under 0 volt duration 350 ms message when PSTN side hang up call, PSTN system will send the 0 volt message.
- Fixed DTMF time – When 38xxA received RFC2833 RTPevent from Asterisk server, it can produce fixed time on (200 ms) DTMF to PSTN, so that PSTN side will detect DTMF from 38xxA easily.
- Disconnect (no rtp received for more than "interval" seconds) –If 38xxA doesn't received RTP packets more than interval seconds, it will auto close channel and disconnects call.
- Cid change –This feature is to replace existing IP line number 1001 of 3804A FXO port with calling line ID number from PSTN switch if it sets to ON. WG3804A will use this calling line ID number to send to remote IP side.

a. disable Caller ID + Cid change: off =>

From:"1001"<sip:1001@192.168.32.253:8088>;tag=c0a820c8-13c4-ca-32037-2

Disable to detect Calling line ID from PSTN switch. Also, disable to change calling ID to be sent to IP side. WG3804A will use this FXO line number to send to remote IP side.

b. enable Caller ID + Cid change: off =>

From:"69835122"<sip:1001@192.168.32.253:8088>;tag=c0a820c8-13c4-ca-32037-2

Enable to detect Calling line ID from PSTN switch. Also, disable to change calling ID to be sent to IP side and using the FXO port IP line number to send to remote IP side..

c. enable Caller ID + Cid change: on=>

From:"69835122"<sip:69835122@192.168.32.253:8088>;tag=c0a820c8-13c4-ca-32037-2

Enable to detect Calling line ID from PSTN switch. Also, enable to change calling ID to be sent to remote IP side. WG3804A will use detected calling line ID number from PSTN switch to send to remote IP side.

- Detect Polarity Reverse –*** "Reverse" it is voltage changed when PSTN side hang up call, PSTN system will send the reverse**.

3.6 Voice Configuration

Users could configure the voice codec or gain level in this web page.
Please get more detail info from the following description. (see figure 3.10)

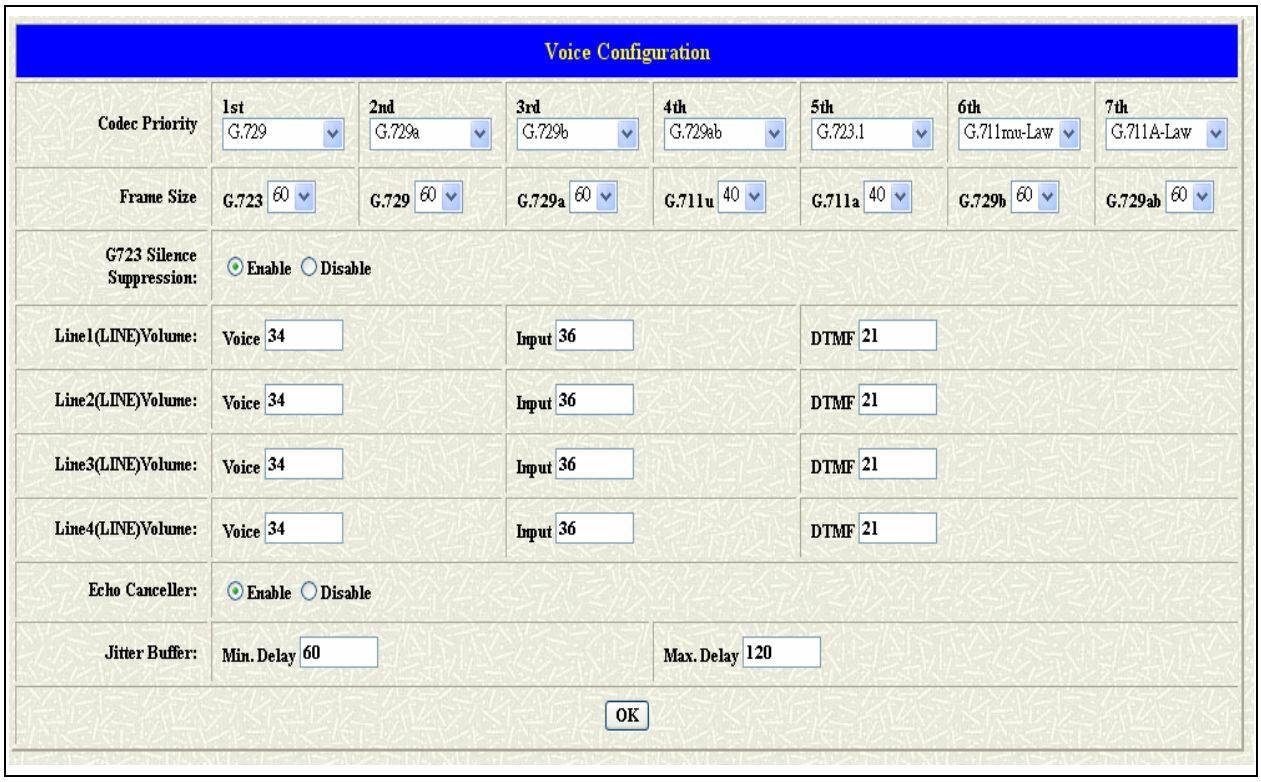


Figure 3.6: System Configuration

- Codec Priority – This could help users configure the codec priority for using. Please pay more attentions about the firmware you use. If the firmware only support G.729 series codec, the G.723 codec in this table will be useful.
- Frame Size – To configure the packet size for the codec that users want.
- G723 Silence Suppression – To enable the VAD and CNG function for the G.723 codec.
- Line1 (TEL 1) Volume – To configure the output gain (voice), input gain (input) and DTMF gain (DTMF) of the first FXS port.
- Line2 (LINE 1) Volume – To configure the output gain (voice), input gain (input) and DTMF gain (DTMF) of the first FXO port.
- Line3 (TEL 2) Volume – To configure the output gain (voice), input gain (input) and DTMF gain (DTMF) of the second FXS port.
- Line4 Volume – To configure the output gain (voice), input gain (input) and

DTMF gain (DTMF) of the second FXO port.

- Echo Canceller – To enable or disable the echo cancellation function.
- Jitter Buffer – To configure the Min or Max delay for the Jitter Buffer. The min is from 0ms and the max is 150ms.

A large jitter buffer causes increase in the delay and decreases the packet loss. A small jitter buffer decreases the delay but increases the packet loss. The size of the jitter buffer depends on the condition of the network, which varies with time. Typically the packet loss should be less than 10% for a good quality of speech.

3.7 Tone Setting

The Tone Setting is for the Tone detecting. The call will be dropped if the pattern of the tone from PSTN side is as same as the pattern in the disconnect tone table. The same result for the Ring Back Tone. User could define the pattern of the disconnect tone if the disconnect tone from PSTN side is not the standard tone. (see figure 3.7)

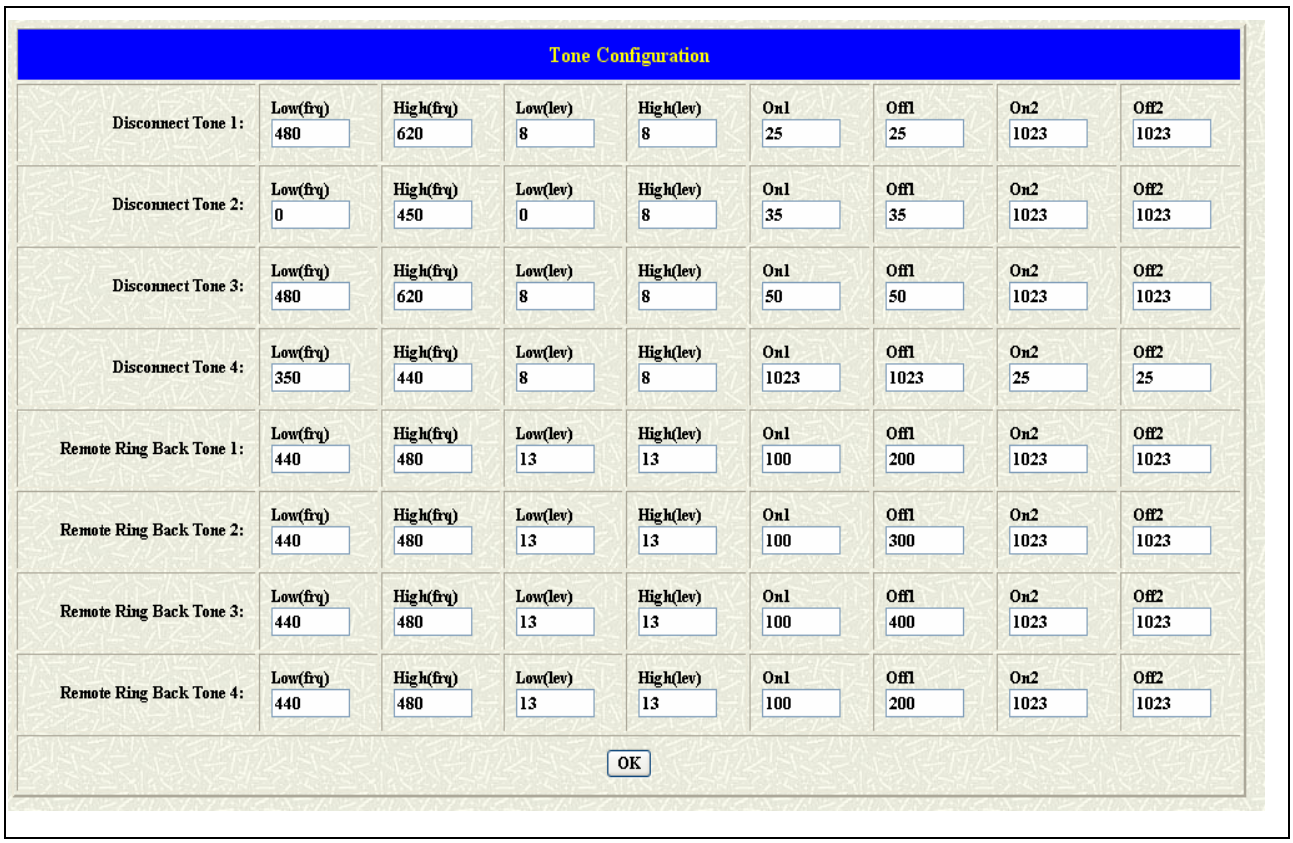

Tone Configuration
Disconnect Tone 1: Low(frq) 480 High(frq) 620 Low(lev) 8 High(lev) 8 On1 25 Off1 25 On2 1023 Off2 1023
Disconnect Tone 2: Low(frq) 0 High(frq) 450 Low(lev) 0 High(lev) 8 On1 35 Off1 35 On2 1023 Off2 1023
Disconnect Tone 3: Low(frq) 480 High(frq) 620 Low(lev) 8 High(lev) 8 On1 50 Off1 50 On2 1023 Off2 1023
Disconnect Tone 4: Low(frq) 350 High(frq) 440 Low(lev) 8 High(lev) 8 On1 1023 Off1 1023 On2 25 Off2 25
Remote Ring Back Tone 1: Low(frq) 440 High(frq) 480 Low(lev) 13 High(lev) 13 On1 100 Off1 200 On2 1023 Off2 1023
Remote Ring Back Tone 2: Low(frq) 440 High(frq) 480 Low(lev) 13 High(lev) 13 On1 100 Off1 300 On2 1023 Off2 1023
Remote Ring Back Tone 3: Low(frq) 440 High(frq) 480 Low(lev) 13 High(lev) 13 On1 100 Off1 400 On2 1023 Off2 1023
Remote Ring Back Tone 4: Low(frq) 440 High(frq) 480 Low(lev) 13 High(lev) 13 On1 100 Off1 200 On2 1023 Off2 1023
OK

Figure 3.7: Tone Setting

- Disconnect Tone – Users could put the correct pattern of the disconnect tone in this table. The call will be dropped if the tone from PSTN side is match with these patterns. Users could have four tables for the disconnect tone.
- Remote Ring Back Tone – User could adjust this table to help the FXO gateway to detect the Remote Ring Back Tone. There could be four tables for the configuration.

3.8 Phone Book

The Phone Book configuration is only support the gateway in Peer-to-Peer mode. Please refer the chapter 2 about the Peer-to-Peer mode. (see figure 3.8)

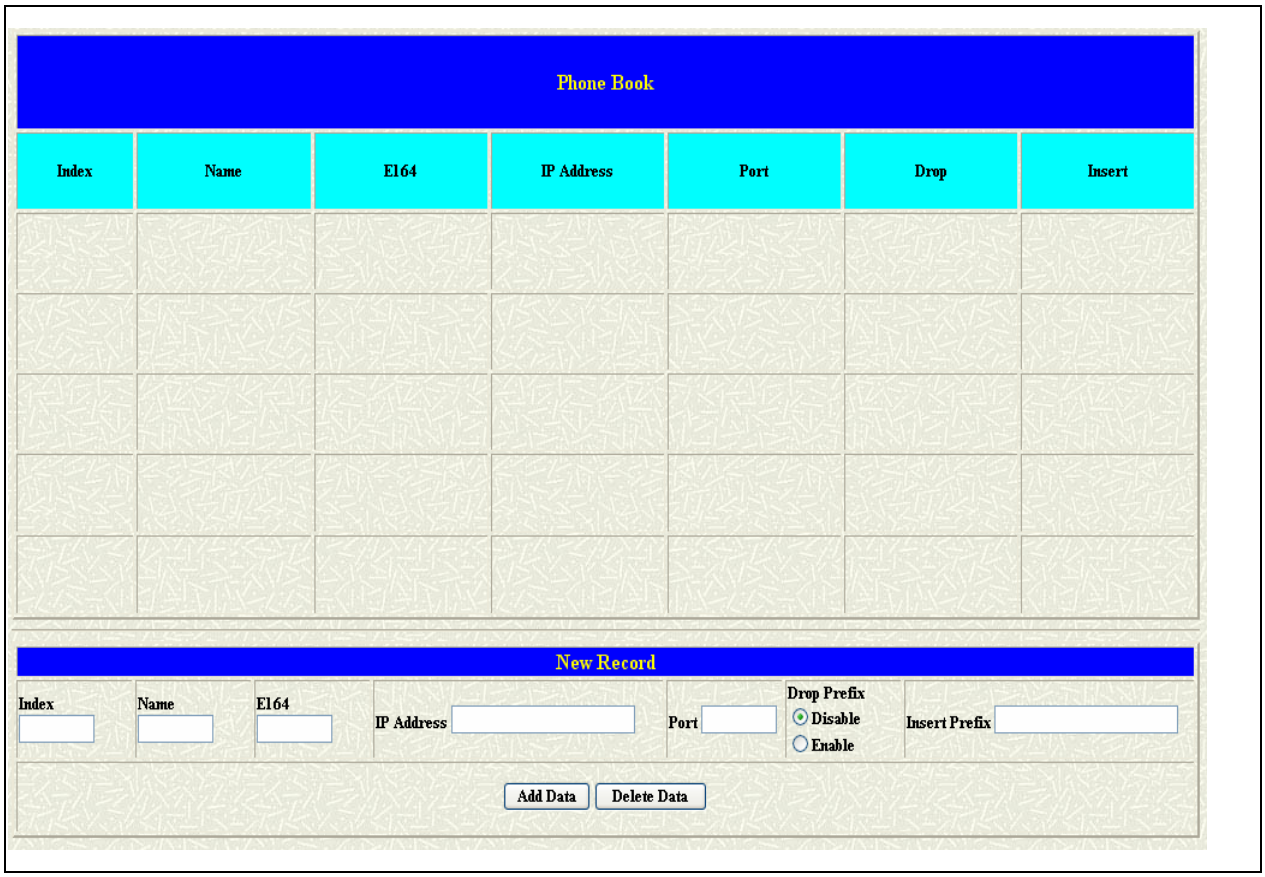


Figure 3.8: Phone Book

- Index – The list number of the Phone Book.
- Name – The name for this contact number.

- E164 – The dialing number for the calling side.
- IP Address – The destination IP address for this phone number.
- Port – The call signal port of the destination.
- Drop – Support the drop function. Enable is for enable this drop function; Disable is for disable this drop function. The Drop Prefix will drop the E164 number, which you had configured in the E164 table.
- Insert – Support the insert digits function.

1. It will be the drop function if user enable the Drop Prefix function and put nothing into the Insert Prefix table.
2. It will be the insert function if user disable the Drop Prefix function and put the digits into the Insert Prefix table.
3. It will be the replace function if user enable the Drop Prefix function and put the digits into the Insert Prefix table.

3.9 Prefix Configuration

The Prefix function is using the drop and inserts function (see figure 3.9).

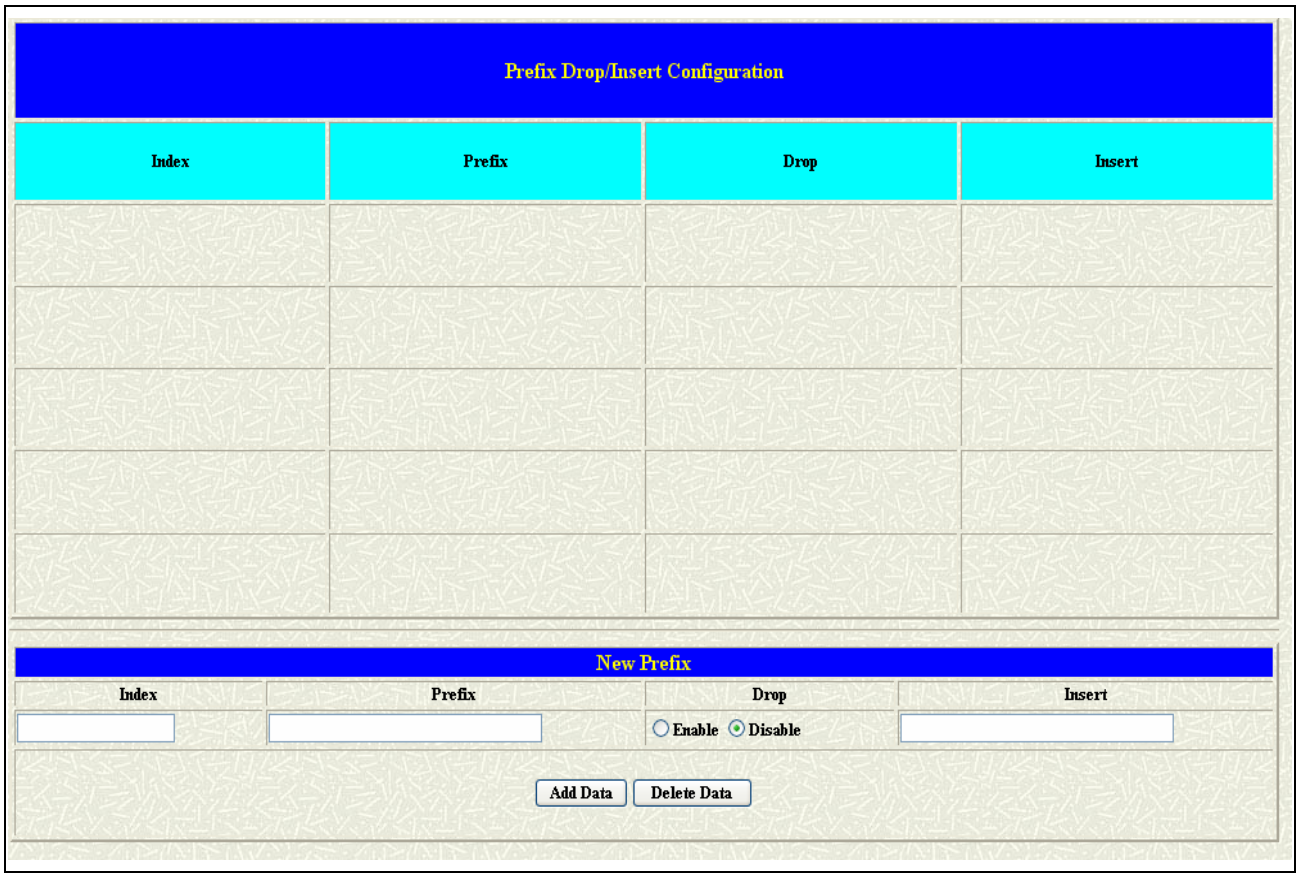


Figure 3.9: Prefix configuration

- Index – The list number of the Phone Book.
- Prefix – The prefix number of the whole numbers that could be into this gateway
- Drop – The drop function. Enable this function by the Enable button; Disable this function by the Disable button.
- Insert – The insert function. Users could enter the digits that you want to insert in this number.
- Add Data – Press this button if users fill the entire information table above.
- Delete Date – If users want to delete the record from the table, enter the index number first and press this button. The record will be deleted.

This function is just like the Phone Book configuration. All the numbers into this gateway will check out the prefix table first and find out the destination in the Routing Table.

There is an example about the configuration, please follow up these steps.

1. Press the Prefix Configuration button to enter the configuration table (see figure 3.9)
2. Enter the index number. Put the prefix numbers you will dial in the prefix table, enable (disable) the drop function and enter the numbers you want to insert (see figure 3.10)

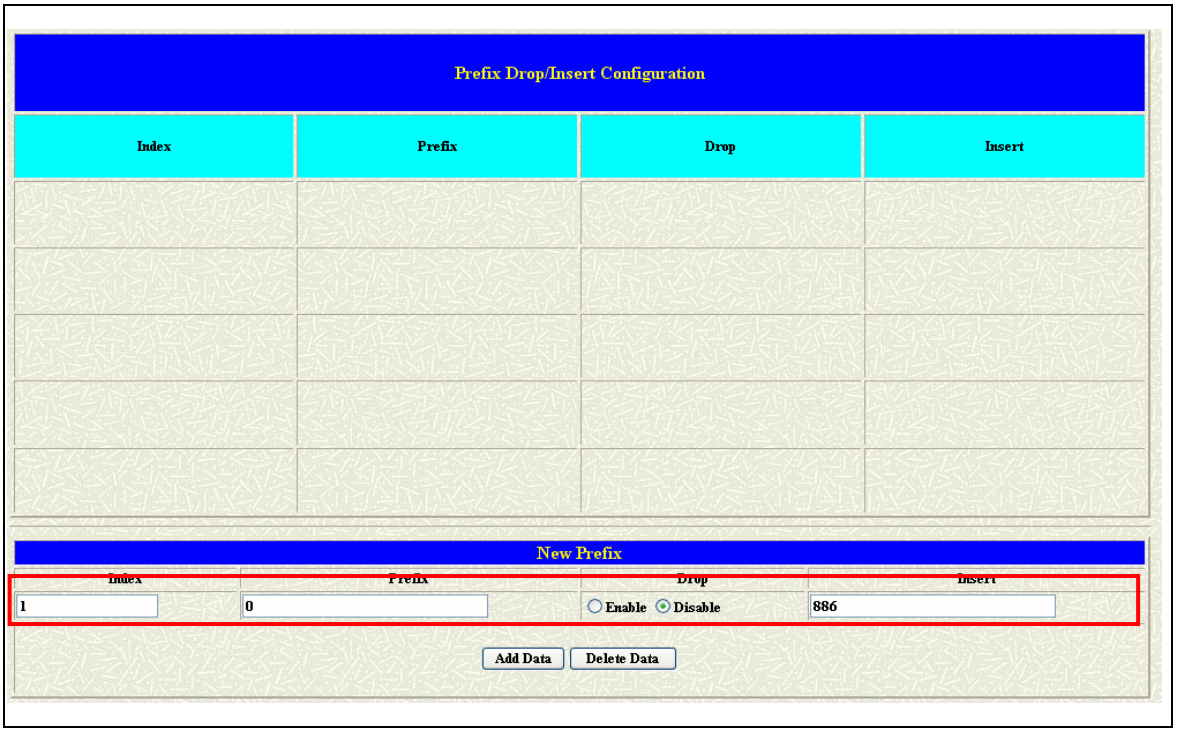


Figure 3.10: Configure the Prefix Table

The usage is as same as the drop, insert function of the Phone Book.

Input (Prefix)	Drop	Insert	Output
100	Disable	X	100
200	Disable	0	0200
300	Enable	X	X
400	Enable	500	500

3. Press the Prefix Configuration button to reload the configuration table (see figure 3.11)

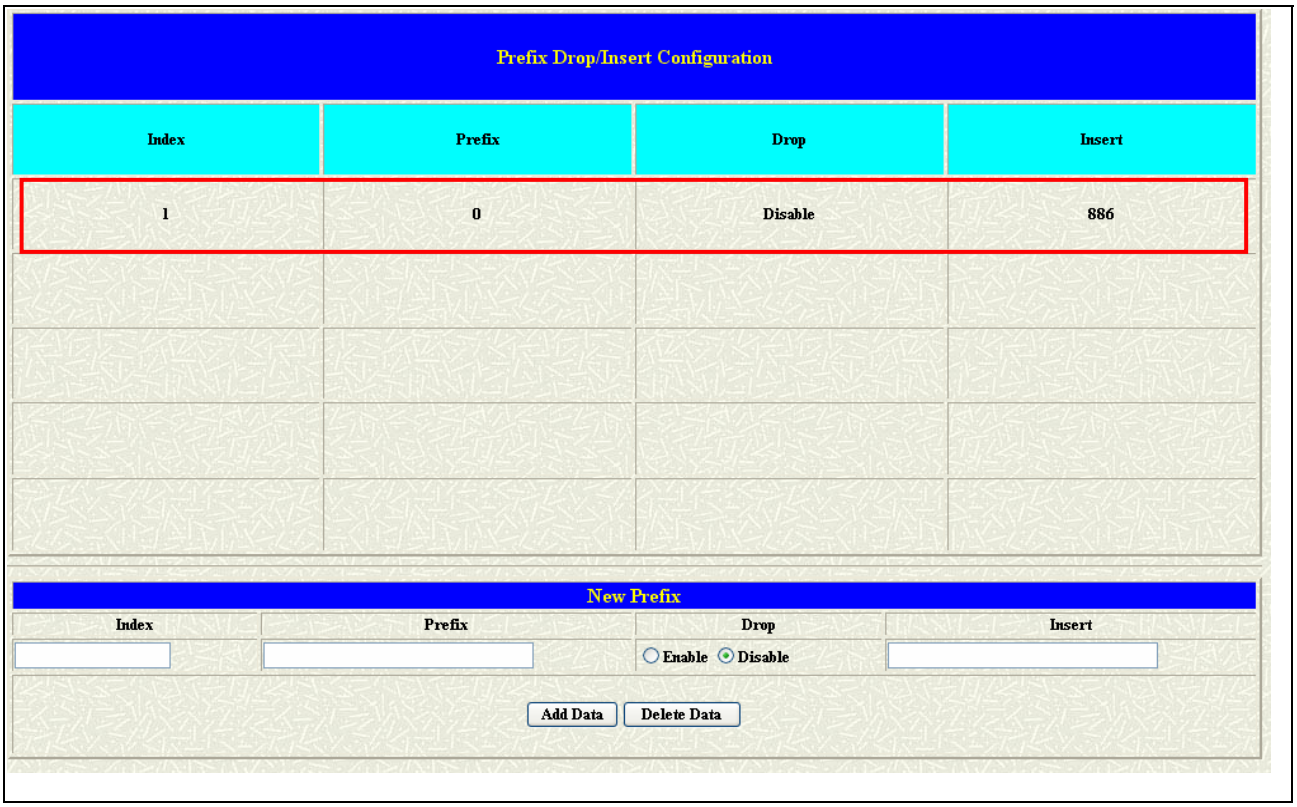


Figure 3.11: Show the added table

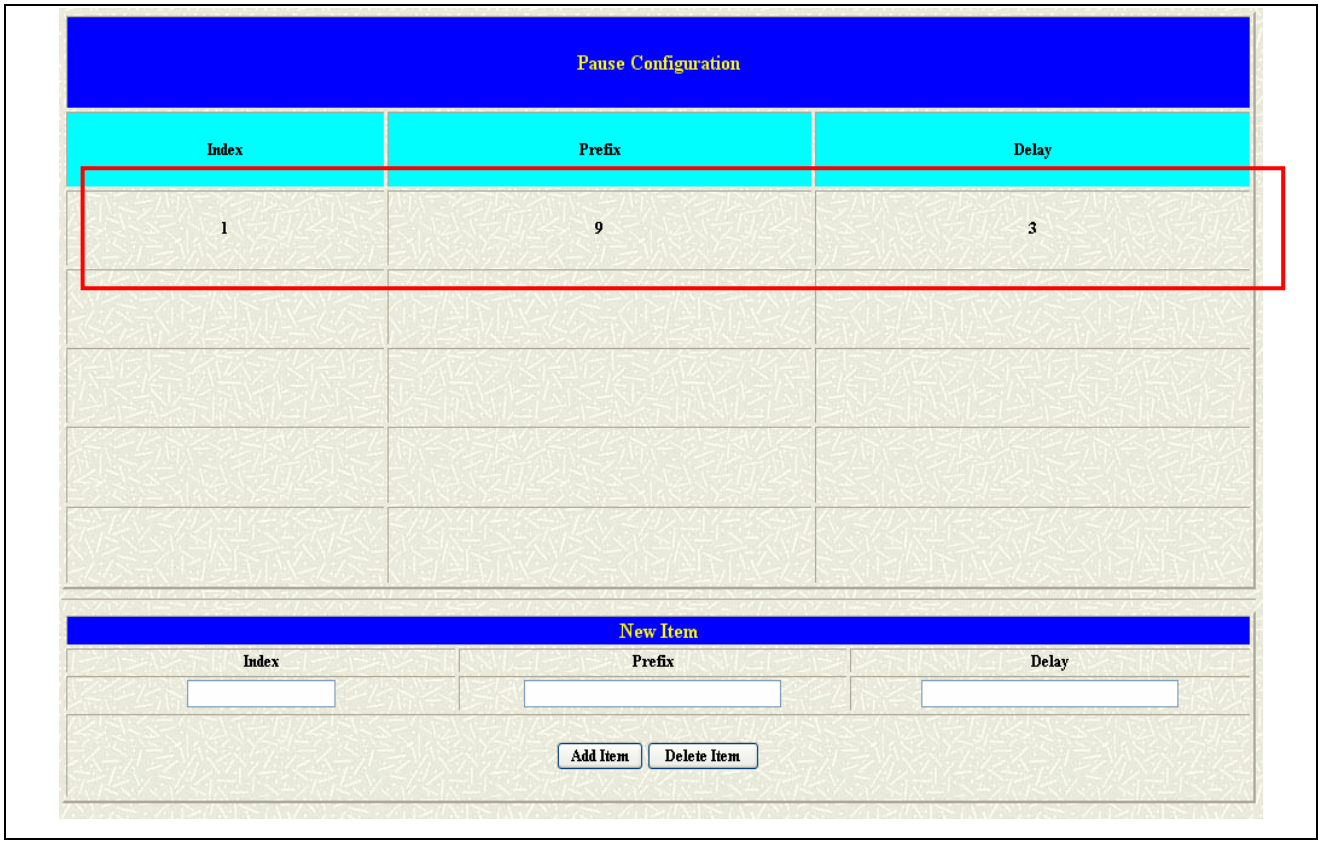
4. Please Commit it and Reboot the system if the configuration is finished.

3.10 Pause Configuration

Pause function is using to after FXO receives a specific prefix number. It will dial the specific prefix number first, then waiting some seconds that you set and continue dial the last number.

For example:

When FXO receives 9123 invite message, FXO will dial 9 first, then waiting 3 seconds and continue dial 123 to PSTN side.



3.11 Routing Table

Routing Table is a rule to define the destination of the calls you make. You could define the rules by the number you dial or by the ports. The Routing Table button will show you the configuration table (see figure 3.11).
In fact, there are two directions of the incoming calls (from IP and FXO side). The explanation of the default routing is as below:

The location with	The location with	The explanation

the incoming calls	the destination	
IP (Default)	Fxs	The destination will be the FXS port when the calls from the IP side without any define rules.
Fxo (Default)	IP	The destination will be the IP side when the calls from the FXO port without any define rules.

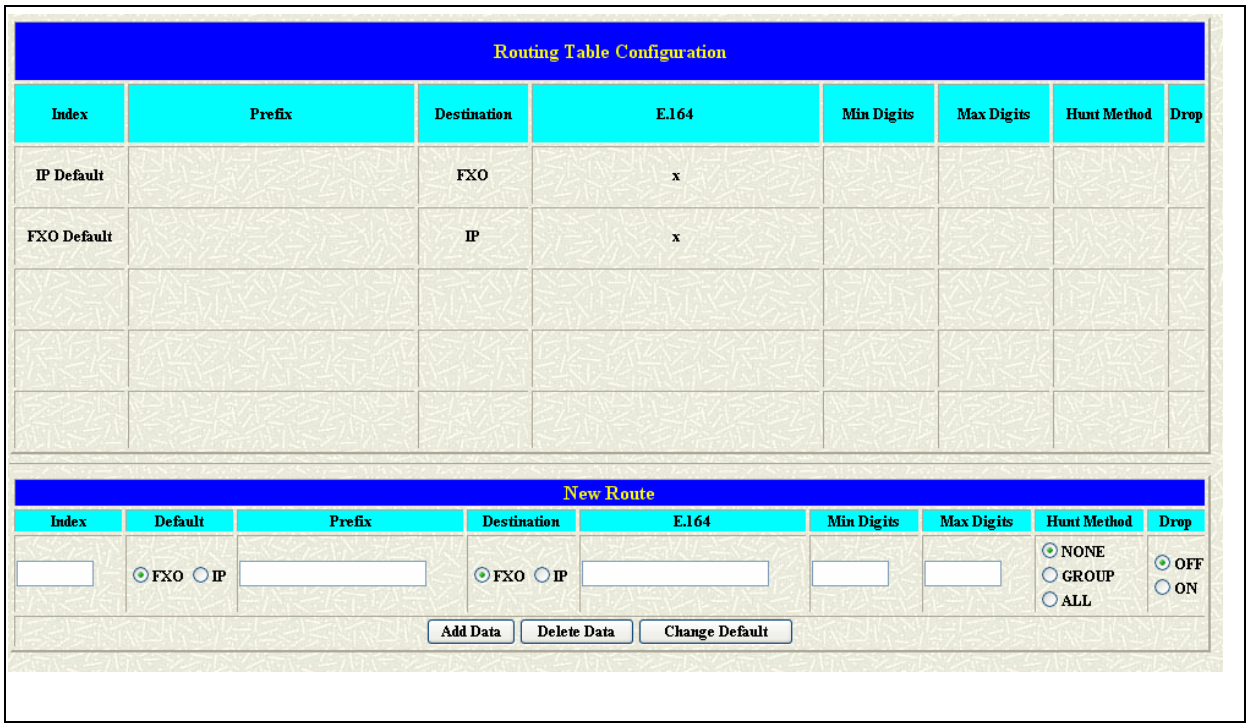


Figure 3.12: Routing Table Configuration

- Index – The list number of the Route Table.
- Default – For change the default setting. Users have to pick the direction for the default setting changed.
- Prefix – The prefix number for the dialed digits. The call will be followed this route table if the prefix number was matched.
- Destination – To decide the destination for this route table.
- E.164 – The E.164 number of the destination.
- Mini Digits – The mini digits requirement for this route table.
- Max Digits – The max digits requirement for this route table.
- Hunt Method – Enable the Hunt Group function and pick up the hunt type.

- NONE: Disable the Hunt Group function.
- GROUP: The Hunt Group function will working for the same group. User could configure the group in the Line Configuration table.
- ALL: The Hunt Group will working for the same type.

◆ Drop – Enable it, it will drop the prefix number.

None – Disable this function
Group – The call will search other ports to be the destination with the same group if the origin destination is busy.
All – The call will search other ports to be the destination with the same type if the origin destination is busy.
About the Group setting, Please get more info from the Line Configuration.

3.12 FXO Password

You will get the IVR if you make calls from PSTN side. The IVR will ask you the password you set, and you could make other calls to IP side if the password you type is correct. Please press the FXO Password button to configure the password (see figure 3.13)

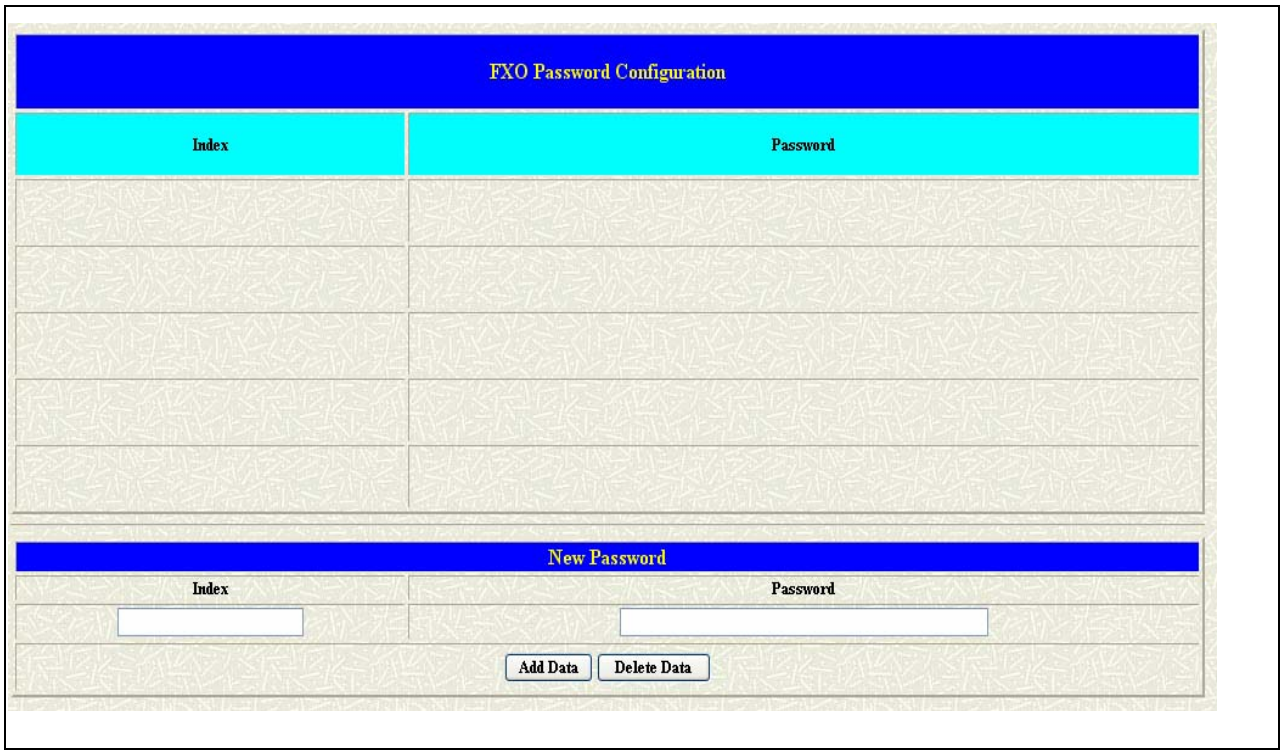


Figure 3.13: FXO Password

◆ Index – The number of this table.

◆ Password – The password you set.

This function is only for the calls from the PSTN side. It's not ready for the IP side as so far.

3.13 IP packet TOS

The Type of Service should be worked with the network router. The router will check all the packets if it support the TOS function. There is a field in the packet for the TOS value. This WEB is for users to configure these values to make the packets with the correct values for the TOS service from the gateway. (see figure 3.14)

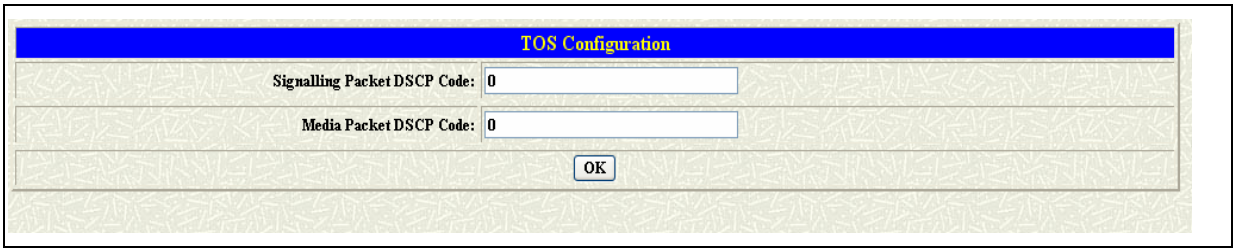


Figure 3.14: ToS

According to the RFC 1349 document, the TOS value as following:

1000 – minimize delay
0100 – maximize throughput
0010 – maximize reliability
0001 – minimize monetary cost
0000 – normal service

These values are the Binary format. Please change to the Decimal and put these values in to the correct table.

3.14 Password Configuration

There are two-login accounts in this unit. One is the account root another is administrator. The default setting for these two accounts are empty. Users

could define the passwords for these two accounts. Please get more info from the following description. (see figure 3.15)

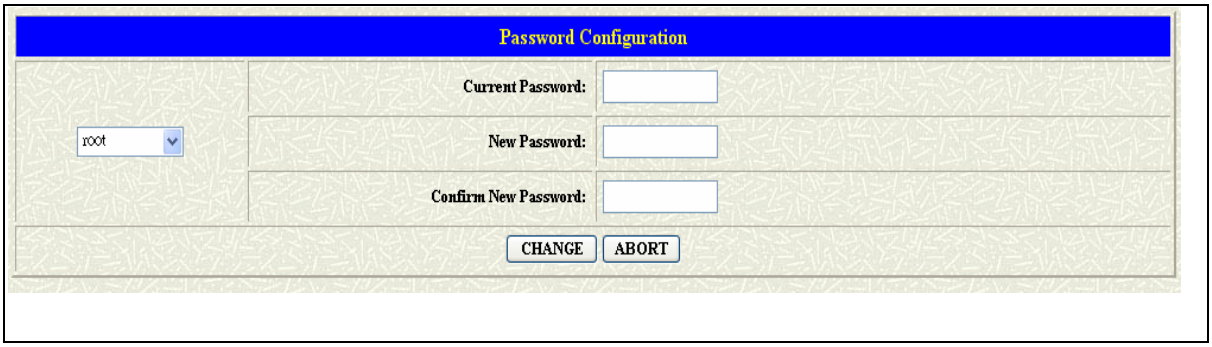


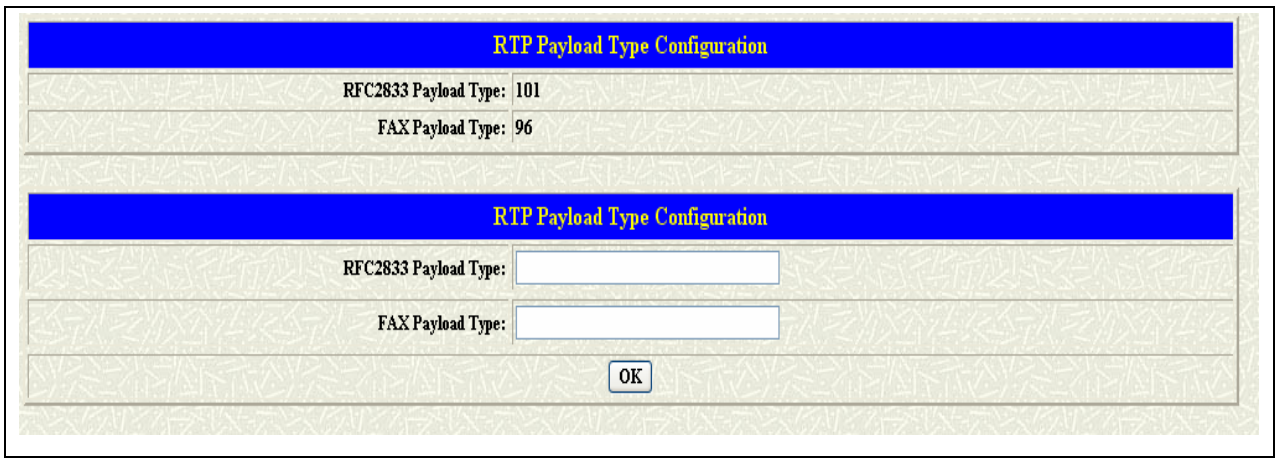
Figure 3.15 Password Configuration

- root – The password for the root account.
- administrator – The password for the administrator account. This account couldn't upgrade the 2M and boot rom file.
- Current Password – Enter the original password for the account.
- New Password – Enter the new password for the account.
- Confirm New Password – Enter the new password again.
- Change – This button will make the configurations saved and next time login will need the new password.
- Abort – Abort the configuration of the password changing.

Please remember the password you configure for the account. It will be more difficult to access it if you forgot the password.

3.15 RTP Payload Type configuration

set RTP2833 RTPevent payload type and FAX Payload type. The two value must be different each other.



3.16 ROM Upgrade

User could update the firmware just by the web configuration interface. There are two types for the upgrading procedure. One is using the TFTP server, another is using the FTP server. Please follow the step to update the gateway firmware version. (see figure 3.16)

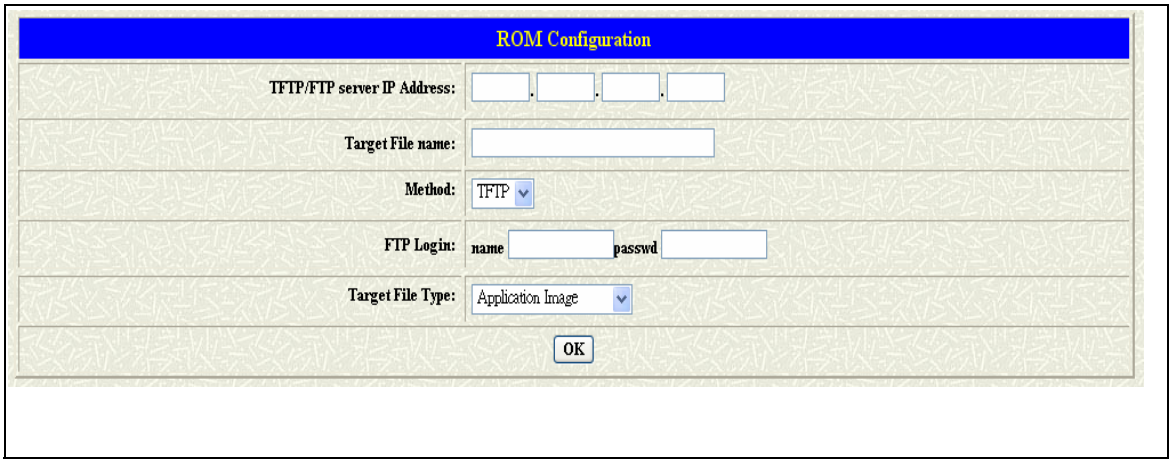


Figure 3.16 ROM Upgrade

- TFTP/FTP server IP Address – Put the ip address of the TFTP or FTP server in this table.
- Target File name – Put the target file name in this table.
- Method – There are two upgrade methods for the upgrade procedure. One is TFTP and another is FTP. Please change to correct method for the upgrading.
- FTP Login – Please enter the login name and password for the FTP

upgrade method. This is necessary if user change the method to the FTP.

- Target File Type – Please pick up the correct file type for upgrading. If the file name and the file type is unconformable, the upgrade procedure will be failed.
- OK – Press the OK button if all the info above are correct. The unit will start to download the firmware file from the TFTP or FTP and write to the flash after the downloading.

【Updating the firmware by the FTP server】

1. Pick up the “Rom Upgrade” button to enter the upgrading web page and switch to the FTP method. (see figure 3.17)

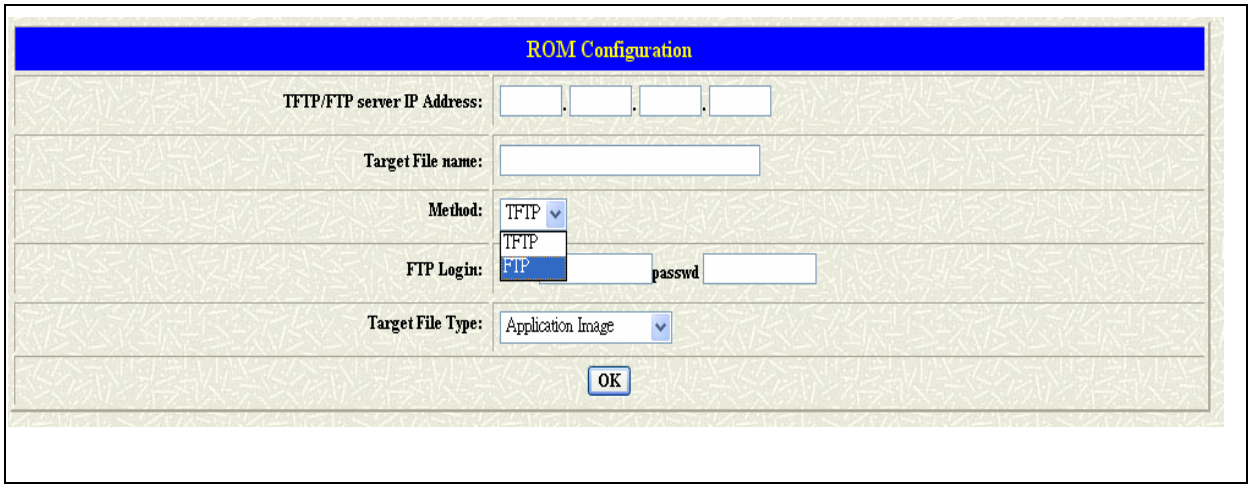


Figure 3.17 ROM Upgrade for FTP

2. Key in the IP address, the login name, password of your FTP server and the correct file name, file type. (see figure 3.18)

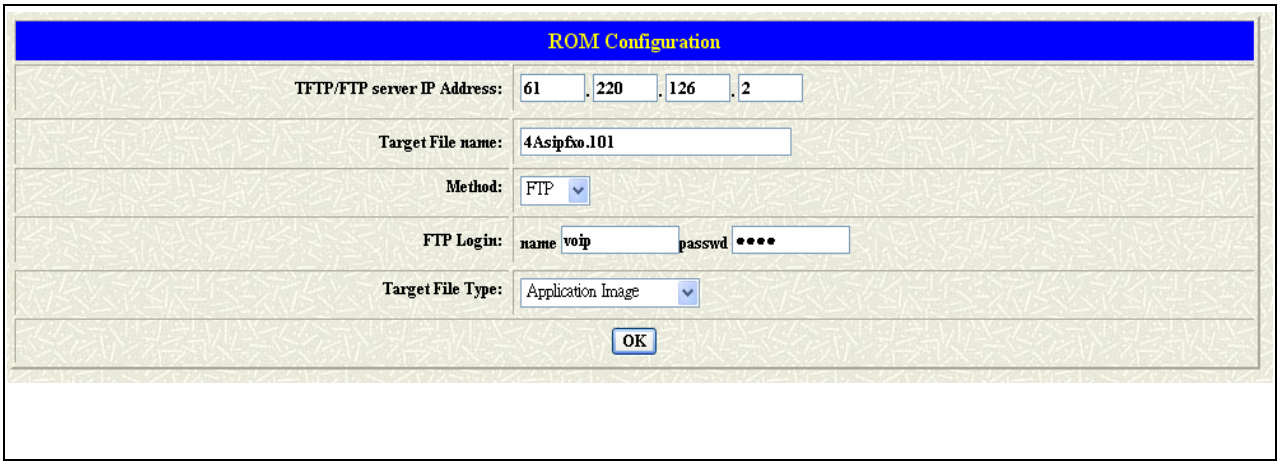


Figure 3.18: FTP information

Please pay more attentions about the red blank. The Target File Type has to be matched with the Target File name. Please put the correct info about the Target file in this table.

4. Please press the “Reboot System” button to make it reboot.

【Updating the firmware by the TFTP server】

1. Downloading the TFTP program from our web site and install it first. Executing the TFTP program before you want to use the TFTP upgrade method.
2. Pick up the “Rom Upgrade” button to enter the upgrading web page and switch to the TFTP method. (see figure 3.19)

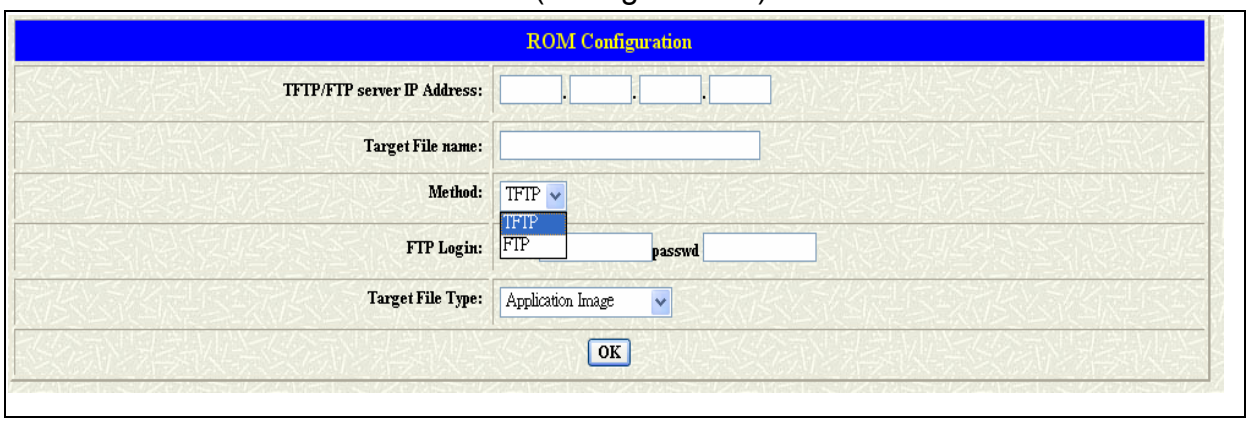


Figure 3.19: ROM Upgrade for TFTP

3. Key in the IP address of your TFTP server, pick up the file type for your upgrade file and the correct file name for upgrading. (see figure 3.20)

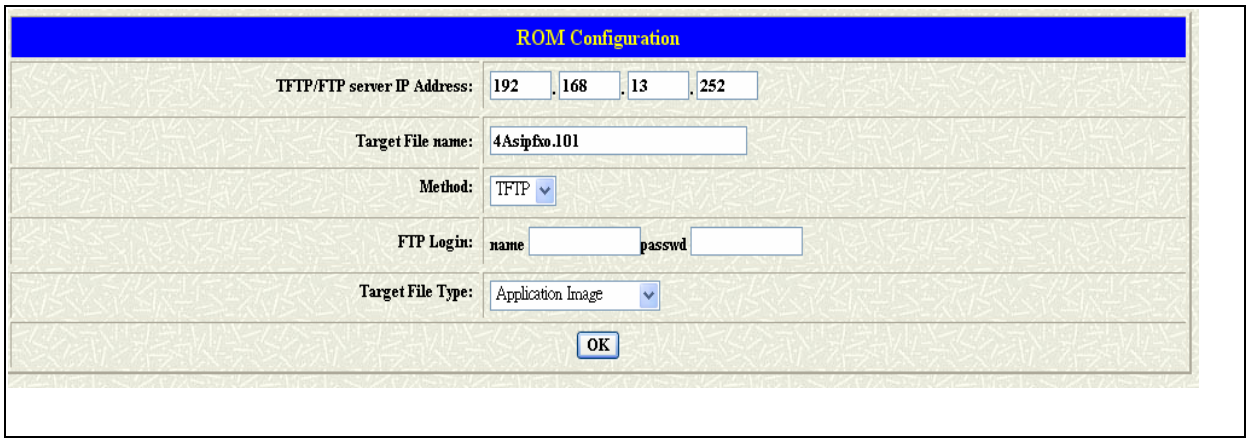
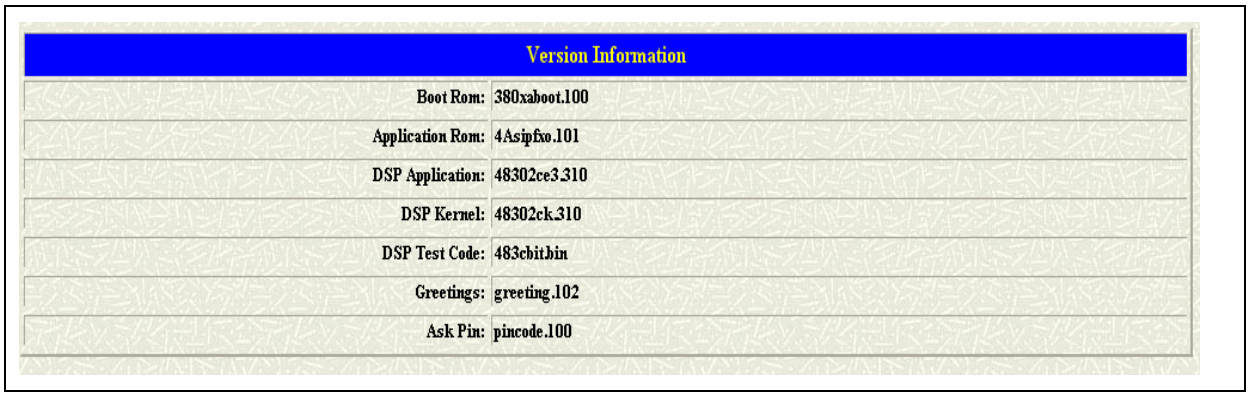


Figure 3.20: TFTP information

4. Press the OK button to start the upgrade procedure.

5. Please press the “Flash Clean” button when the procedure is finished.
6. After pressing the “Flash Clean” button, please press the “Reboot System” button to make it reboot.

3.17 Version Information



3.18 Flash Clean

Users could make all the configurations back to the default setting by this button. The password of the account and the networking configuration couldn't be back to the default setting by this command. (see figure 3.21)

Figure 3.21: Flash Command

3.19 Commit

This web page could save the configurations if users change some configurations. This is necessary for users change the configurations. (see figure 3.22)

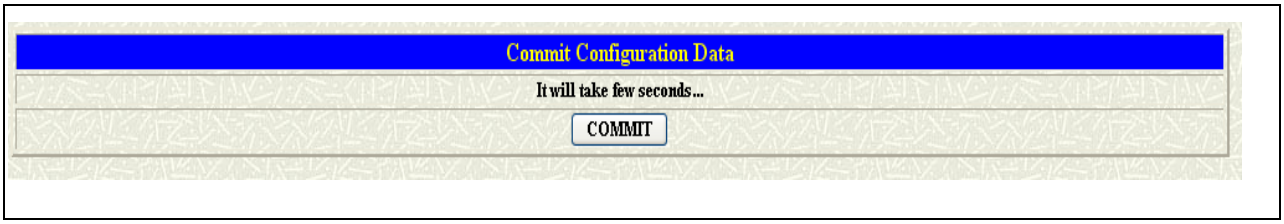


Figure 3.22: Commit the data

3.20 Reboot System

This web page will restart the whole system. This is the necessary step for the changing the configurations and makes it executed. (see figure 3.23)

Figure 3.23: Reboot System

4. Command List

4.1 Hyper Terminal Setting

A terminal emulator is needed when using RS-232 port to configure Gateway. There are kinds of terminal emulator software. Here, we use Microsoft HyperTerminal to depict how to set up terminal emulator:

1. Execute the *Hyper Terminal* program, and then the following windows will pop-up on the screen. (START – Program files – Accessories – Communication – Hyper Terminal)

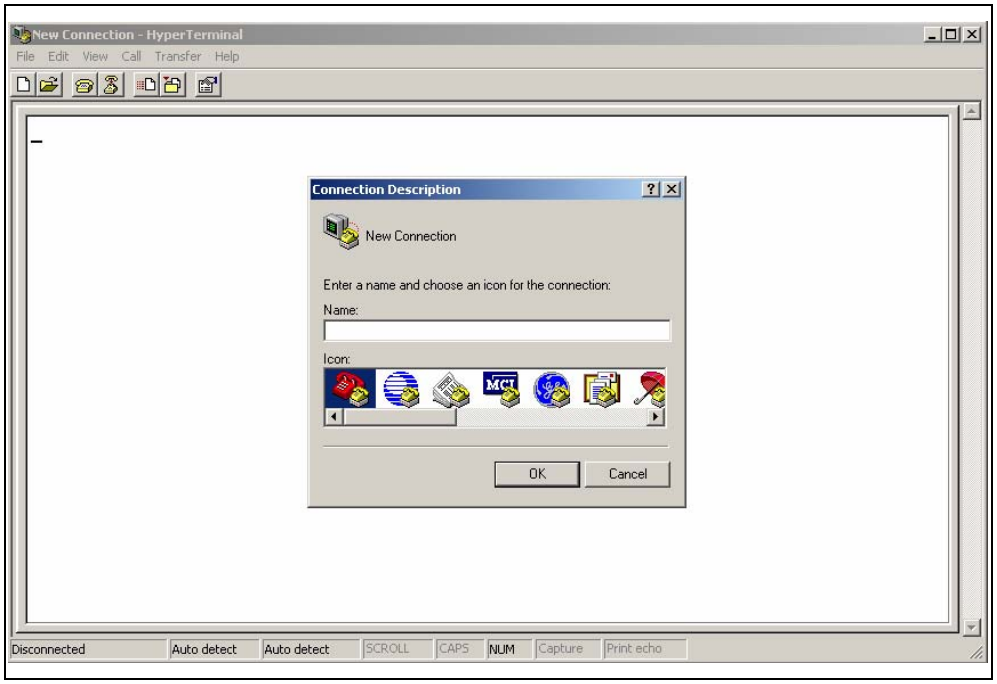


Figure 4.1: Hyper Terminal

2. Define a name such as ‘voip’ for this new connection.

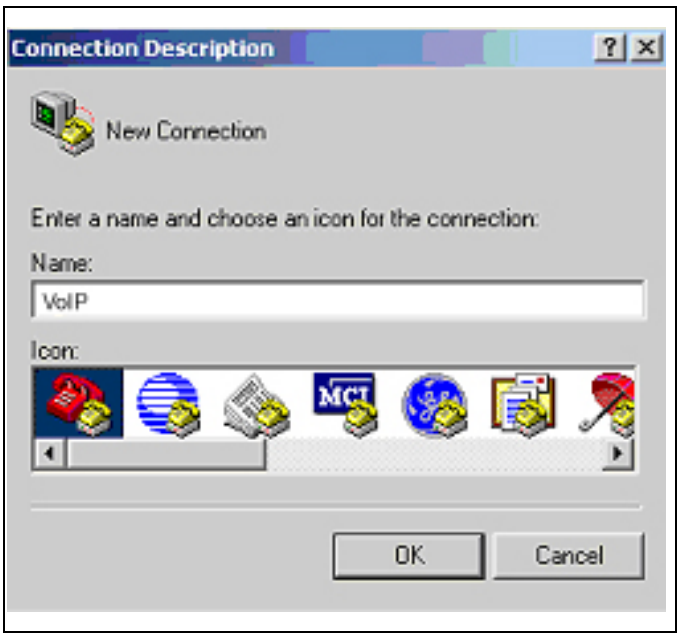


Figure 4.2: Edit the name for the connection

3. After pressing OK button, the next window appear, and then choose ***COM1/2 Port***, which you are going to use.

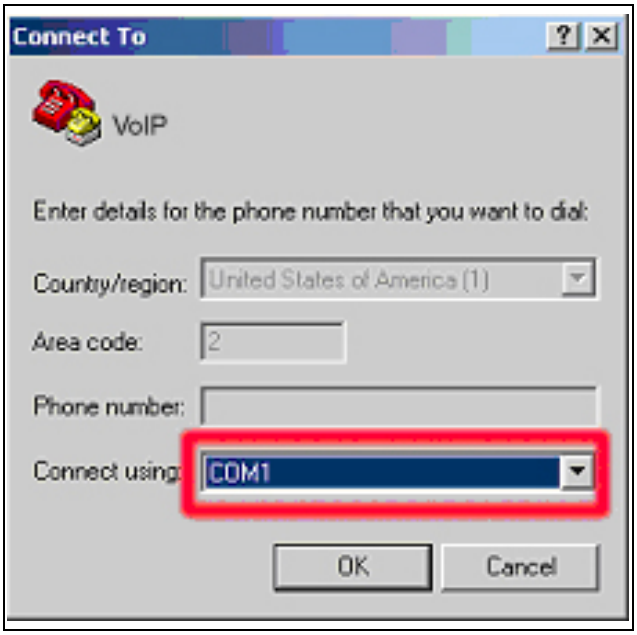


Figure 4.3: Pick up the right interface to use

4. Configure the COM Port Properties as following:
   - Bits per second: 9600
   - Flow control: None

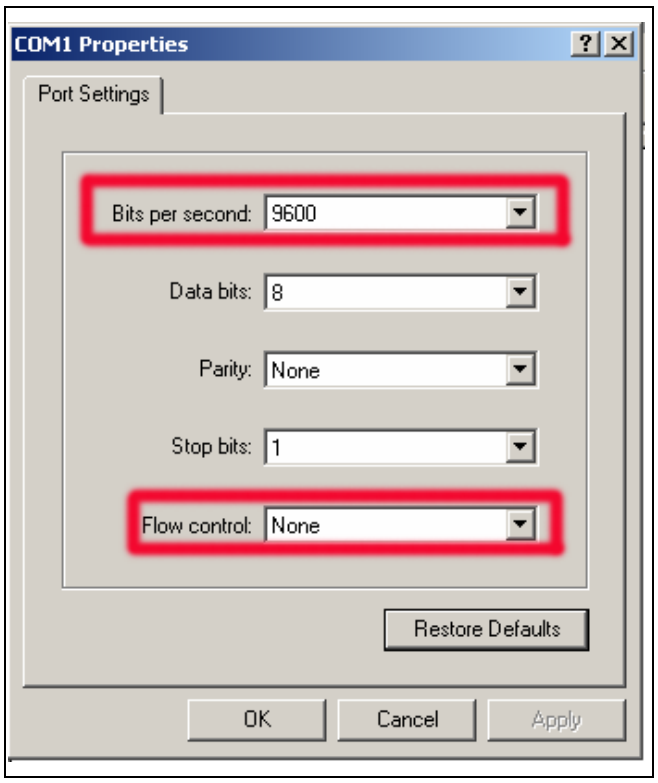


Figure 4.4: Configure the right Bps and control

5. Press ‘OK’ button, and then start to configure Gateway.

4.2 Command List

4.2.1 [help]

Type **help** or **man** or **?** to list all the available command.

```
usr/config$ ?
help            help/man/? [command]
quit            quit/exit/close
debug           show debug message
reboot          reboot local machine
flash           clean configuration from flash rom
commit          commit flash rom data
ifaddr          internet address manipulation
time            show current time
ping            test that a remote host is reachable
sysconf         System information manipulation
sip             SIP information manipulation
security        Security information manipulation
line            Line information manipulation
route           Routing information manipulation
prefix          Prefix drop/insert information manipulation
pause           FXO Pause information manipulation
pbook           Phone book information manipulation
voice           Voice information manipulation
tone            Setup of disconnect tone and remote ring back tone
fxopwd          Setup of FXO password
record          Record voice for greeting and ask pin code
support         Special Voice function support manipulation
tos             IP Packet ToS (Type of Service)values
pt              DSP payload type configuration and information
rom             ROM file update
passwd          Password setting information and configuration
auth            Set configuration items for "administrator" user.

usage: help [command]

usr/config$
```

4.2.2 [quit]

Type **quit** will quit the configuration mode. And turn back to login prompt.

```
usr/config$ quit
Disconnecting...
login:
```

Note: *It is recommended that type the "**quit**" command before you leave the console. If so, the unit will ask password again when next user connects to console port.*

4.2.3 [debug]

Open debug message will show up specific information while FXO GateWay is in operation. After executing the debug command, it should execute command **debug -open** as well. One example is demonstrated below.

```
usr/config$ debug -add sip vp
usr/config$ debug -open
```

Parameters Usage:

-status	Display the enabled debug flags.
-add	Add debug flag.
	-- sip : sip related information
	-- vp : voice related information
-delete	Remove specified debug flag.
-open	Start to show debug messages.
-close	Stop showing debug messages.

4.2.4 [reboot]

After **commit** command, type **reboot** to reload FXO GateWay in new configuration. The procedure is as below:

```
usr/config$ reboot
Attached TCP/IP interface to cpm unit 0
Attaching interface lo0...done
AC4804[0] is OK
AC4804[1] is OK
AC4804[2] is OK
```

Successful

Initialize OSS libraries...OK!
open stack successful
cmInitialize succeed!
GK mode selected.

login:

4.2.5 [flash]

This command will clean the configuration stored in the flash rom and reboot FXO GateWay in factory default setting.

Parameter Usage:

-clean clean all the user-defined value, and reboot FXO GateWay in factory default mode.

Note: It is recommended that use "flash –clean" after application firmware id upgraded.

Warning: Once users execute **flash –clean,** all the configurations of FXO GateWay will be cleaned. This can only be executed by user who log in with **root**

4.2.6 [commit]

Save changes after configuring the FXO GateWay.

---

usr/config$ commit

This may take a few seconds, please wait....
Commit to flash memory ok!
usr/config$

---

*Note: Users should use **commit** to save modified value, or they will not be activated after system reboot.*

4.2.7 [ifaddr]

Configure and display FXO GateWay network information.

usr/config$ ifaddr

LAN information and configuration
Usage:
ifaddr [-print]\|[-mode used]\|[-sntp mode [server][-cmcenter ipaddress]]
ifaddr [-ip ipaddress] [-mask subnetmask] [-gate defaultgateway]
ifaddr [-dns index [dns server address]]
ifaddr [-id username][-pwd password][-http http port]

-print Display LAN information and configuration.
-ip Specify ip address.
-mask Set Internet subnet mask.
-gate Specify default gateway ip address
-mode Set ip client service(0=FIX IP, 1=DHCP, 2=Pppoe).
-sntp Set SNTP server mode and specify IP address.
-dns specify IP address of DNS Server.
-timezone Set local timezone.
-id Connection user name for pppoe.
-pwd Connection password for pppoe.
-http Http port.

Note:
SNTP mode (0=no update, 1=specify server IP, 2=broadcast mode).
Example:
ifaddr -ip 210.59.163.202 -mask 255.255.255.0 -gate 210.59.163.254
ifaddr -mode 1
ifaddr -sntp 1 210.59.163.254
ifaddr -dns 1 168.95.1.1

usr/config$

Parameters Usage:

-print	print out current [ifaddr] settings and status
-ip	assign IP address for Gateway
-mask	assign internet subnet mask

-gate	assign IP default gateway
-mode	Switch the network type (0 = Static IP; 1 = DHCP mode 2 = PPPoE mode)
-sntp	Simple Network Time Protocol (1 = ON; 0 = OFF) When SNTP function is activated, users have to specify a SNTP server as network time source. An example is demonstrated below:
-autodns	set specify the way to obtain DNS Server (0:Manual/ 1:Auto)
-dns	configure the IP address for the DNS server
-timezone	set local time zone according to GMT

---

usr/config$ ifaddr -sntp 1 10.1.1.1

10.1.1.1 stands for SNTP server's IP address.

-id	To configure the pppoe connection account for the pppoe connection.
-pwd	To configure the pppoe connection password for the pppoe connection.
-http	Change the http port. User can change default HTTP port (80) to another one for security or NAT application.

4.2.8 [time]

When SNTP function of Gateway is enabled and SNTP server can be found as well, type **time** command to show current network time.

---

usr/config$ time
Current time is THU JAN 01 05:29:23 1970

Please make sure about that the SNTP server was configured before you used this command.

4.2.9 [ping]

Use **ping** to test whether a specific IP is reachable or not.

For example: if 192.168.1.2 is not existing while 192.168.1.254 exists.

Users will have the following results:

```
usr/config$ ping 192.168.1.2
no answer from 192.168.1.2
usr/config$ ping 192.168.1.254

PING 192.168.1.254: 56 data bytes
64 bytes from 192.168.1.254: icmp_seq=0. time=5. ms
64 bytes from 192.168.1.254: icmp_seq=1. time=0. ms
64 bytes from 192.168.1.254: icmp_seq=2. time=0. ms
64 bytes from 192.168.1.254: icmp_seq=3. time=0. ms
----192.168.1.254 PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms)    min/avg/max = 0/1/5
usr/config$
```

4.2.10 [sysconf]

This command displays system information and configurations.

```
-------------------------------------------------------------------------------------------
-------
usr/config$ sysconf

System information and configuration
Usage:
  sysconf [-idtime digit][-keypad dtmf]
            [-rba digit][-eod digit][-bl digit]
            [-localrbt digit]
            [-ring on_time off_time]
  sysconf -print

  -print            Display system overall information and configuration.
  -idtime           Inter-Digits time.(1~10 sec)
  -keypad           Select DTMF type: 0=In-band,
                                      1=RFC2833
```

2=INFO.

-callerid	*CallerId Type .(Caller type, 0: none, 1: FSK, 2: ETSI)*
-dtmfstart	*Drop DTMF CallerId Start Start Symbol From PSTN.*
-dtmfend	*Drop DTMF CallerId End Symbol From PSTN.*
-ring	*The ring time for ring detection.(Uint:ms)*
-delay	*The FXO dial DTMF delay.(1~9)(Uint:s)*
-rba	*the number of ring times before answer.(1~5)*
-eod	*End of dial.(Enable:1 / Disable:0)*
-silence	*Silence Detection.(0=Disable, 1=Enable)*
-connect	*Auto connect time.(0=Disable, for 1~65535 sec)*
-onhook	*Auto ON-HOOK if detect reverse.(Enable:1 / Disable:0)*
-ivr	*General IVR in FXO.(Enable:1 / Disable:0)*
-sipping	*Enable the SIP-Ping function(0: OFF, 1: ON)*
-fixeddtmf	*Enable Fixed DTMF Tone (0: OFF, 1:ON)*
-cdrop	*Current Drop detect (Disable:0 / Enable:1)*
-flashtime	*Flash time setting.(1-200, 1->10ms)*
-sendinfo	*Flash/Reverse Switch. (0: OFF, 1: ON)*

Example: sysconf -ring 500
usr/config$

Parameters Usage:

-print print out all current settings

-idtime set the duration(in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, it will dial out all number pressed. (1-10 seconds)

-keypad DTMF replay type. When value is "0", Gateway will transfer DTMF signal via In-Band type, "1" via RFC2833 type. "2" is SIP info method. Users can adjust the value according to various applications.

1. *number (instead of Line number of FXO Line)+ PSTN number to make a call to PSTN side connected with FXO Line.*
2. *After gateway-prefix-drop function is enabled, user must remember to re-configure line number of FXS Line, because line number of FXS Line must remove prefix*

number. For example, origin line number of FXS line is 1001, prefix is 100, since prefix number will be drop, once gateway has incoming call 1001, after drop gateway prefix 100, it will search line number "1". So line number must be set as "1".

-callerid the FXO can display the PSTN CallerID.
there are three type callerID type, FSK / ETSI / DTMF. So you must make sure which callerID type PSTN system will send.

-dtmfstart Drop the start DTMF symbol of callerID.

-dtmfend Drop the end DTMF symbol of callerID.

-ring ring time for ring detection(in ms). When Gateway has incoming call from PSTN side to FXO port, Gateway will determine it is a ring but not noise only if it is longer than this ring time.

Note:

In Taiwan the ring time of PSTN usually is 1000ms, so if user set ring time longer that 1000ms, FXO port may not be able to pick up the call from PSTN side

-delay When FXO port has an incoming call from IP side and signal connection is established, it will wait the dial tone from PSTN or PBX. But sometimes the dial tone from PBX is too late so some errors will occur. Now user can use this command to change the time waiting for dial tone.

-rba When the calls from the PSTN side, FXO port will off hook if the ring time is matched with this number.

-eod It will transfer the DTMF in "#" if users disable the end of dial function. Users have to press the key pad in "#" if the end of dial function is enabled.

-silence Enable or Disable silence detect function.

-connect The unit will send the connect message- 200OK to the IP side automatically. This function just supports the one-stage dialing function.

-onhook Enable or Disable FXO reverse detect function. Welltech FXO has two ways to release FXO port, one is disconnect tone detect, another is PSTN reverse. This command can let user enable or disable FXO reverse detect function.

-ivr User can disable FXO default greeting by this command,

using dial tone insteaded.

-sipping This function is for some special platform, such as Nortel proxy server. The FXO will send SIP Ping message to server before registering or making call.

-fixeddtmf This function is producing fixed time on(200 ms) DTMF tone to PSTN system, when FXO receives RFC2833 RTPevent packets.

-cdrop **"Current Drop", it is a loop current drop signal to FXO port message when remote PSTN side calling party hang up the call, the local PSTN switch will drop the loop current to local FXO gateway. This feature is widely used in North and South America PsTN switch or office PABX to inform called party that this call was disconnected.**

-flashtime The FXO gateway can support [flash transparent] mode if the user in IP side press flash button. The FXO gateway will generate a flash signal to the PBX or PSTN. You can change the flash time here, default is 30 (300ms).

-sendinfo This function must be used with 35XX, because 35XX will send specific SIPINFO message to FXO, FXO will produce related flash time to PSTN system.

4.2.11 [sip]

Phone Book function allows users to define their own numbers, which mapping to real IP address. It is effective only in peer-to-peer mode. When adding a record to Phone Book, users do not have to reboot the machine, and the record will be effective immediately.

---

usr/config$ sip

SIP stack information and configuration
Usage:
sip [-mode pxmode]
sip [-px address] [-outpx address] [-prefix prefixstring] [-line number]
[-expire t1]
sip -print

-print Display SIP stack information and configuration.
-mode Configure as Proxy mode or Peer-to-Peer mode or Gateway mode.
-px Proxy server address. (Proxy IPv4 address or Proxy dns name)
-pxport Proxy server port. (the port of proxy)
-outpx OutBound Proxy server address. (IPv4 address or dns name)
-prefix Specify prefix string, use it when UserID contains alphabets (if UserID uses numerals, specify as null)
-line1 Line 1 is E.164 number of L1.
-line2 Line 2 is E.164 number of L2.
-line3 Line 3 is E.164 number of L3.
-line4 Line 4 is E.164 number of L4.
-pbsearch Search phone book 0:off/1:on.
-lmedia Late Media 0:off/1:on.
-cidchange CID Change in From Header 0:off/1:on.
-expire The relative time after which the message expires(0 ~ (2^31-1))
-port SIP local UDP port number (2326~65534), Default: 5060
-rtp RTP port number (2326~65534), Default: 16384
-busy busy response message, Default: 486

Example:

sip -px 210.59.163.171 -line1 70 -line2 71

usr/config$

Parameter Usages:

-print print current h323 related settings

-mode alternatives for proxy or peer-to-peer mode (1=proxy mode; 0=peer-to-peer mode). If users select proxy mode, a valid proxy is needed when Gateway is in operation.

---

usr/config$ sip –mode 0 (peer to peer mode)

-px to assign the ip address of the proxy when Gateway is in proxy mode.

-pxport define the proxy port for the registration or call.

-outpx define the out bound proxy for the endpoints.

-prefix this will be prefix the alphabets before the sip line number.
-line1 assign FXO Line1 number.
-line2 assign FXO Line2 number.
-line3 assign FXO Line3 number.
-line4 assign FXO Line4 number.

Note:

User can also set "x" in line number to disable the port. If the port is disabled, it can only receive calls but not calling out.

Note:

1. For the 4 ports FXO units, it will show the line3 to line 4. For the 2 ports FXO units, it will show only line1 to line2.
2. No matter in Proxy or P2P mode, user only needs to dial line number to reach local port. For example, in P2P mode, user wants to dial from FXS TEL1 to FXO Line1, only need to dial number of line2.

-pbsearch enable/disable phone book search function under Proxy Mode. If user enabled this function, SIP-Phone will search dialed number in phone book to see if there is any matched table before send to Proxy server, and if there is a matched data in phone book, SIP-Phone will make call to related IP address.

-lmedia Some platform will send and reply the SDP in "SIP 200OK" and "ACK". But the FXO gateway can only send and reply SDP in SIP Invite and SIP 200OK. Enable this function will push the FXO gateway to send and reply SDP in "SIP 200OK" and "ACK.

-cidchange The FXO will set the PSTN caller ID in the SIP "display name" field if the FXO can detect the PSTN caller ID successfully. But some platform can only will also confirm the calling number via the "SIP FROM header". Enable this function will set the PSTN caller number instead of the FXO local number. Please disable this function if you are not sure what you are doing.

-expire It just like the TTL function in H323, the gateway will make sure the registration is success or not for a period times.

-port define the local sip port for this gateway.

-rtp to allocate RTP port range—NOT RECOMMENDED. This

may be used when RTP port range conflicts with Firewall policy. (each port of Gateway use 2 RTP ports)

-busy User can define a special response message to server if all slots are busy but get an IP incoming call.

4.2.12 [security]

This is the authentication for the SIP account.

---

usr/config$ security

Secuirty information and configuration
Usage:
security [-name username] [-password password]
security -print

-print Display system account information and configuration.
-line Specify which line number you want to set the account.
-name Specify user name.
-password Specify password.
Example:
security -line 1 -name kkk -password 12345

usr/config$

Parameter Usages:

-print print out all current settings of security.

-line the line number, which you want to define the security info

-name the name is as same as the SIP number.

-password the password for the authentication if it is the necessary for the proxy.

4.2.13 [line]

This command is for configure each line parameters of Gateway.

---

usr/config$ line

Gateway line information and configuration
Usage:

line -config number [hunt number][hotline number]
line -print Gateway line information.
hunt Hunting group.
hotline Hot line configuration.
Example:
line -config 1 hunt 1 hotline 1003

usr/config$

Parameter Usages:

-print print out all current settings of line

-config determine which line to configure

-hunt set hunting group flag of each line. User can assign different hunt group number represent different hunt group. For example, if user assigns FXS TEL1 as hunt group 1, and FXS TEL2 as hunt group 2, they will be determined as 2 different groups. On the other hand, if user assigns FXS TEL1 as hunt group 1, and FXS TEL2 as hunt group 1 too, when having incoming call to FXS TEL1, which is busy, this call will be route to FXS Line2.

Note: *FXO Lines and FXS TELs are treated as 2 different groups, so even they are in the same hunt group, call will only be routed to the same FXS or FXO Lines.*

-hotline set hotline table.The Hotline Mode is applied in limited two channels. Users just make calls into the FXO port and gateway will automatically dial out a phone number. In the other hand, user will hear ring back tone or dial tone immediately depended on configurations of destination device.

Note: This function can both work in Proxy or P2P mode.

Proxy Mode Usage:

Set gateway under proxy mode.

Create a Hotline table with "***line*** " command.

---

usr/config$ line –config 1 hotline 1001

In this example means: if users make calls into the FXO Line1, gateway will automatically dial out “1001”.

P2P Mode Usage:

Set gateway under P2P mode.

Create phone book table with “***pbook*** ” command.

Create a Hotline table with “***line*** ” command.

---

usr/config$ pbook –add name sipfxs ip 10.1.1.1 e164 1001

usr/config$ line –config 1 hotline 1001

In this example means: if users make call into the FXO Line1, gateway will automatically dial out IP address of “1001”.

4.2.14 [route]

This command is to set routing table for Gateway.

usr/config$ route

Routing table information and configuration
Usage:
route -add [prefix number][dst number][e164 number]
[min number][max number][hunt number]
route -delete index
route -modify index [prefix number][dst number][e164 number]
[min number][max number][hunt number]
route -ip [dst number][e164 number]
route -fxo [dst number][e164 number]
route -print Routing table information.
prefix The prefix of dialed number.
dst Destination port(FXO:1/IP:2).
e164 Destination e164 number(when destination is FXO).
min Min digits.(0 ~ 255)
max Max digits.(0 ~ 255)
hunt Hunt method for busy forward(NONE:0/GROUP:1/ALL:2).
Example:
route -add prefix 100 dst 1 e164 1001 min 1 max 3 hunt 1

route -ip dst 1 e164 1001
route -fxo dst 2 e164 x
route -modify 1 prefix 100 dst 2 e164 1001 min 1 max 3 hunt 1
route -delete 1

usr/config$

Parameter Usages:

-print	print out all routing table information
-add	add a routing rule in routing table. User can add less than 50 rules. (***route –add prefix "prefix number" dst "destination port type" e164 "SIP number of port" min "minimum digits needed" max "maximum digits can't be exceeded"***)
-delete	delete a routing rule in routing table (***route –delete "index of routing rule"***)
-modify	modify a routing rule in routing table. (***route –modify "index of routing rule" prefix "prefix number" dst "destination port type" e164 "SIP number of port" min "minimum digits needed" max "maximum digits can't be exceeded"***)
-ip	create routing table for incoming call from IP side. (***route –ip dst "destination port type" e164 "SIP number of port"***)
-fxo	create routing table for incoming call from FXO Lines. (***route –fxo dst "destination port type" e164 "SIPnumber of port"***)
prefix	prefix of dialed number
dst	destination port, 0 means FXS TELs, 1 means FXO Lines, 2 means IP side, x means no determinate number.
e164	destination SIP number. This only need to be set when routed port is FXS TELs or FXO Lines to determine which port will this call be routed to.
min	minimum digits needed.
max	maximum digits needed.
hunt	set hunt method for busy forward. 0 means no hunting, 1 means hunting method follows the rule of ***[line]***, 2 means hunting method is to hunt between all ports in the same type, for example, destination port is FXS TEL will hunt in all FXS

TELs, destination port is FXO Lines will hunt in all FXO Lines.

Usage Example:

1. ***route –add prefix 100 dst 1 e164 1001 min 1 max 3 hunt 1***
   This command means if gateway has incoming call's prefix number is 100, and total digits is between 1 to 3, this call will be routed to FXO port 1001, and if port 1001 is busy, call will be routed to another FXO port.
2. ***route –ip dst 1 e164 1002***
   This command means incoming call from IP side will be routed to FXO Line of number 1002.
3. ***route –fxo dst 2***
   This command means incoming call from FXO Lines will be routed to IP side.

Note:

(1) When destination is IP side, SIP number doesn't need to determine. (Ex. route –fxo dst 2)

(2) If user doesn't want to determine a specific port to route, SIP number must set as "x". (Ex. route –ip dst 1 e164 x)

4.2.15 [prefix]

This command is for make rules for drop or insert prefix digits.

---

usr/config$ prefix

Prefix drop/insert information and configuration
Usage:
prefix -add [prefix number][drop number][insert digits]
prefix -delete index
prefix -modify index [prefix number][drop number][insert number]
prefix -print Prefix drop/insert information.
prefix The prefix of dialed number.
drop Drop prefix(Enable:1/Disable:0).
insert Insert digits.
Example:
prefix -add prefix 100 drop 1 insert 2000

prefix -add prefix 100 drop 1
prefix -add prefix 100 drop 0 insert 200
prefix -delete 1
prefix -modify 1 prefix 100 drop 0 insert 300

usr/config$

Parameter Usages:

-add add a rule to drop or insert prefix digits of incoming call.(***prefix –add prefix "prefix number" drop 0/1 insert "insert number"***)

-delete delete a rule to drop or insert prefix digits of incoming call. (***prefix –delete prefix "prefix number"***)

-modify modify a rule to drop or insert prefix digits of incoming call. (***prefix –modify prefix "prefix number" drop 0/1 insert "insert number"***)

prefix set which prefix number to implement prefix rule.

drop enable or disable drop function.If this function is enabled, Gateway will drop prefix number on incoming call.

insert set which digit to insert on incoming call.

4.2.16 [pause]

Pause function allows users define a prefix for FXO, it usually apply to one-stage-dialing.

For example, the FXO port is connect to a PBX, when an incoming call from IP side, users will hear a dial tone from PBX. If they want to dial to a PSTN, they must press a special code and wait 1~2 seconds for the PSTN dial tone. But in one-stage-dialing application, the FXO will not wait for the dial tone and it will dial immediately. Now user can define a special prefix, so if FXO detect the prefix, it will wait a moment then keep dialing.

---*u*

usr/config$ pause
Prefix drop/insert information and configuration
Usage:
pause -add [prefix number][delay number]
pause -delete index
pause -modify index [prefix number][drop number][insert number]
pause -print Prefix drop/insert information.

prefix The prefix of dialed number.
delay delay time(second).

Example:
pause -add prefix 100 delay 1
pause -delete 1
pause -modify 1 prefix 101 delay 0

usr/config$

Parameter Usages:

-add add a new record to pause function. When adding a record, users have to specify ***prefix*** and ***delay*** seconds to complete the command.
-delete delete a record to pause function.
-print print out current contents of Pause function.

4.2.17 [pbook]

Phone Book function allows users to define their own numbers, which mapping to real IP address. It is effective only in peer-to-peer mode. When adding a record to Phone Book, users also have to reboot the machine, and the record will be effective immediately.

----------*u*

usr/config$ pbook

Phone book information and configuration
Usage:
pbook [-add [name string][e164 number][ip address]
[drop digit][insert number]]
[-modify number [name string][e164 number][ip address]
[drop digit][insert number]]
[-delete number]
pbook -print

-print Display phone book information and configuration.
-add Add new phone book record)
-delete Delete phone book record
-modify Modify phone book record.

name : 1 ~ 10 characters.
e164 : 1 ~ 10 digits.
ip : IP adress.
port : 1024 ~ 65535.
drop : 0:Disable/1:Enable.
insert : 1 ~ 10 digits.

Example:

pbook -add name test e164 1234 ip 192.168.1.10 drop 1 insert 5678
pbook -delete 1
pbook -modify 1 name test e164 5678 ip 192.168.1.10 drop 0

usr/config$

Parameter Usages:

-print — print out current contents of Phone Book. (***pbook -print***) Users can also add *index number*, from 1 to 100, to the parameter to show specific phone number. (Ex. ***pbook –print 1***)

Note: <index number> means the sequence number in phone book. If users do request a specific index number in phone book, Gateway will give each record a automatic sequence number as index.

-add — add a new record to phone book. When adding a record, users have to specify ***name***, ***ip***, and ***e164*** number to complete the command.

- name — name to represent callee.
- e164 — e.164 number for mapping with IP address of callee
- ip — ip address of callee
- port — call signal port number of callee
- drop — drop e.164 number when dial out. 0 means to keep e.164 number, 1 means to drop e.164 number when dialing out.
- insert — insert digits.(1~10 digits)

-modify — modify an existing record. When using this command, users have to specify the record's index number, and then make the change.

-delete — delete a specific record. "pbook –delete 3" means delete

index 3 record.

PhoneBook Rules:

The e164 number defined in phone book will fully carry to destination. It is not just a representative number for destination's IP Address. In other words, user dial this e164 number to reach destination, destination will receive the number and find out if it is matched to its e164, including Line number in some particular device.

4.2.18 [voice]

The voice command is associated with the audio setting information. There are four voice codecs supported by Gateway.

---

usr/config$ voice

Voice codec setting information and configuration
Usage:
voice [-send [G723 ms] [G729 ms] [G729A ms] [G729B ms] [G729AB ms] [G711U ms] [G
711A ms]]
[-volume line [voice level] [input level] [dtmf level]]
[-nscng [G711U used1] [G711A used2] [G723 used3]]
[-echo used] [-mindelay t1] [-maxdelay t2] [-optfactor f]
voice -print
voice -priority [G723] [G729] [G729A] [G729B] [G729AB] [G711U] [G711A]

-print Display voice codec information and configuration.
-send Specify sending packet size.
G.723 (30/60/90 ms)
G.729 (20/40/60 ms)
G.729A (20/40/60 ms)
G.729B (20/40/60 ms)
G.729AB (20/40/60 ms)
G.711U (20/40/60 ms)
G.711A (20/40/60 ms)
-priority Priority preference of installed codecs.

G.723
G.729
G.729A
G.729B
G.729AB
G.711U
G.711A

-volume Specify the following levels:
voice volume (0~63, default: 30),
input gain (0~63, default: 30),
dtmf volume (0~31, default: 23),

-nscng No sound compression and CNG. (G.723.1 only, On=1, Off=0).

-echo Setting of echo canceller. (On=1, Off=0, per port basis).

-mindelay Setting of jitter buffer min delay. (0~150, default: 90).

-maxdelay Setting of jitter buffer max delay. (0~150, default: 150).

Example:

voice -send g723 60 g729 60 g729a 60 g729b 60 g729ab 60 g711u 60 g711a 60

voice -volume 1 voice 20 input 32 dtmf 27

voice -echo 1 1 1 1

usr/config$

Parameters Usage:

-print print current voice information and configurations.

-send define packet size for each codec. 20/40/60ms means to send a voice packet per 20/40/60 milliseconds. The smaller the packet size, the shorter the delay time. If network is in good condition, smaller sending packet size is recommended. In this parameter, 20/40/60ms is applicable to G.711u/a law, and G.729/G.729A/G.729B/G.729AB codec, while 30/60ms is applicable to G.723.1 codec.

-priority codec priority while negotiating with other SIP device. This parameter determines the listed sequence in SDP message. The codec listed in left side has the highest priority when both parties determining final codec. User can also select the

particular codec without others.

usr/config$ voice –priority g723 (only select this codec)
usr/config$ voice –priority g723 g729 g711u g711a (select four codecs, and g723 is the first choice)

-volume There are three adjustable values. **voice volume** stands for volume, which can be heard from Gateway side; **input gain** stands for volume, which the opposite party hears; **dtmf** volume stands for DTMF volume/level, which sends to its own Line.

Note: level of volume is too high or too low may be result in bad performance while connecting to each other. Please make the value below 35.

-nscng silence suppression and comfort noise generation setting (1 = ON; 0 = OFF). It is applicable to G.723 codec only. An example is demonstrated below:

usr/config$ voice -nscng g723 1

-echo activate each canceler (1 = ON; 0 = OFF).
-mindelay the minimum jitter buffer size. (Default value= 90 ms)
-maxdelay the minimum jitter buffer size. (Default value= 150 ms)

usr/config$ voice –mindelay 90 –maxdelay 150

*Note: be sure to know well the application before you change **voice** parameters because this might cause incompatibility.*

4.2.19 [tone]

This command is basically for FXO ports.

---

usr/config$ tone

Disconnect tone and remote ring back tone configuration
Usage:

tone [num][freqLo][freqHi][freqLoLev][freqHiLev]
[Tone1ON][Tone1OFF][Tone2ON][Tone2OFF]]
tone -print Display tone configuration.
[num] Tone index(1~4:Disconnect tone / 5~8:Remote ring back tone).
Example:
tone -print
tone 1 480 620 8 8 50 50 1023 1023

usr/config$

Parameter Usages:

-print show all tone configuration

[num] tone index. 1~4 is disconnect tone, 5~8 is remote ring back tone.

For FXO ports Gateway must detect disconnect tone to determine when to disconnect the call, so user must set disconnect tone of PBX or PSTN network connected to FXO ports.

When making a call from FXO ports, there are 2 ways to detect cal lee has already picked up the call, one is to detect reverse signal, the other is to detect the termination of ring back tone, so user must set ring back tone of PBX or PSTN network.

(If user doesn't know about the frequency of disconnect tone or ring back tone, please refer to ***[record]*** command to detect frequency.)

For each tone may has 1 set or 2 sets (high and low) of frequencies. If user wants to set 0 in on/off time, please set "1023" represent "0". (ex. ***tone 1 620 480 8 8 50 50 1023 1023***)

(***tone "index of tone" "frequency of high" "frequency of low" "level of high" "level of low" "on time of high" "off time of high" "on time of low" "off time of low"***)

4.2.20 [fxopwd]

This command is for FXO ports.

---

usr/config$ fxopwd

FXO password information and configuration
Usage:
fxopwd -add [passwd number]
fxopwd -delete index
fxopwd -modify index [passwd number]
fxopwd -print FXO password information.
passwd The password.
Example:
fxopwd -add passwd 1234
fxopwd -delete 1
fxopwd -modify 1 passwd 1234

usr/config$

Parameter Usages:

-print	show all FXO password configuration
-add	add 1 set of FXO password
-delete	delete 1 specific set of FXO password
-modify	modify 1 specific set of FXO password
passwd	password

4.2.21 [record]

User can record greeting and askpin file and analyze tone frequency by calling in FXO line of Gateway.

---

usr/config$ record

Recoed greeting voice and ask pin code voice, tone analizes.
Usage:
record -greeting filename
-askpin filename

-tone

Example:

record -greeting greeting.100

record -askpin askpin.100

record -tone

usr/config$

Parameter Usages:

-greeting record greeting file. User must assign a file name for greeting, once record is finished, file recorded will be display in rom –print.

---

usr/config$ record -greeting greeting.test

Please off hook the LINE 1 and press (N) for the next step...

n

Press (R) to start record...

r

Press (S) to stop record...

.....

.....

.....s.....

.....

Press (P) to play the voice or (W) to write to flash or (Q) to quit...

p

w

Please wait a moment...

Write flash ok...

Boot Rom : boot.104

Application Rom : 4fxo.118a

DSP App : 48302ce3.140
DSP Kernel : 48302ck.140
DSP Test Code : 483cbit.bin
Greetings : greeting.test
Ask Pin : askpin.102

q
usr/config$

-askpin record askpin file. User must assign a file name for askpin file, once record is finished, file recorded will be display in rom –print.

---

usr/config$ record -askpin askpin.test

Please off hook the LINE 1 and press (N) for the next step...
n

Press (R) to start record...
r

Press (S) to stop record...
.....
.....
.....s.....
.....
.....

Press (P) to play the voice or (W) to write to flash or (Q) to quit...
p
w

Please wait a moment...
Write flash ok...

Boot Rom : boot.104
Application Rom : 4fxo.118a

```
                DSP App   :   48302ce3.140
             DSP Kernel   :   48302ck.140
          DSP Test Code   :   483cbit.bin
              Greetings   :   test.100
              Ask Pin     :   askpin.test
q
usr/config$
```

-tone analyze tone frequency. Gateway can analyze tone frequency as users provide tone in FXO Line1.

```
usr/config$ record –tone

usr/config$ record -tone


Press (R) to start record...
r
.....
.....
.....
.....
.....
Analizing!! Please wait a moment...
Frequency 1 : 620
Frequency 2 : 474
0.25sec on 0.25sec off
usr/config$
```

The Procedures of recording the disconnect tone

Before you start ：

Two PSTN lines which connected with the Line 1 and Line 2 port.
Configure Peer-to-Peer mode.

Please record the disconnect tone just follow the stage as below ：

1. Please enter the command before you record the disconnect tone：*record –tone*
2. Make a call from PSTN side into Line 2 port.
3. You will get a greeting when the calls enter the gateway.
4. Pease dial the number of the Line 1 port.
5. You will get another dial tone from the PSTN line which was connected with the Line 1 port.
6. Answer the call and make sure the call is connected.
7. Hang up the phone which is from the Line 2 port and you will get the disconnect tone from the Line 1 port.
8. When you get the disconnect tone from the Line 2 port, press <**R**> and <**ENTER**> buttons to start recording the disconnect tone.
9. Please hang up the phone if you get the message as below：*Analizing!! Please wait a moment…*
10. There are three values you will get after analyzing. Please leave the value which is over 1000 Hz, this is not the frequency of disconnect tone.
11. Please put the frequency in the tone table just follow the command：
    tone 4 420 680 8 8 25 25 50 50

【Example-1】

(Make a call from PSTN to FXO port)
usr/config$ record -tone

Press (R) to start record...
(Please make sure that you are already finish the steps 2 ~ 7)
r (Press “Enter” button after you key in “R”)
.....
.....
.....
.....

Analizing!! Please wait a moment...
(You coule hang up the call from PSTN if you get this message)

Frequency 1 : 481

Frequency 3 : 621
0.25sec on 0.25sec off

tone 4 481 621 8 8 25 25 1023 1023
(Put this value in to the tone table)
tone –print

Disconnect tone 1 paramter
Frequency high : 620
frequency low : 480
frequency high level : 8
frequency low level : 8
Tone1 on : 25
Tone1 off : 25
Tone2 on : 1023
Tone2 off : 1023
Disconnect tone 2 paramter
Frequency high : 450
frequency low : 0
frequency high level : 8
frequency low level : 0
Tone1 on : 35
Tone1 off : 35
Tone2 on : 1023
Tone2 off : 1023
Disconnect tone 3 paramter
Frequency high : 620
frequency low : 480
frequency high level : 8
frequency low level : 8
Tone1 on : 50
Tone1 off : 50
Tone2 on : 1023
Tone2 off : 1023
Disconnect tone 4 paramter
Frequency high : 621
frequency low : 481

frequency high level : 8
frequency low level : 8
Tone1 on : 25
Tone1 off : 25
Tone2 on : 50
Tone2 off : 50
(Confirm the values is correct or not)

(Key in the commit and reboot command if you finish the procedures as above)

【Example-2】

(Make a call into FXO port)
usr/config$ record -tone

Press (R) to start record...
(Please make sure that you are already finish the steps 2 ~ 7)
r (Press “Enter” button after you key in “R”)
.....
.....
.....
.....

Analizing!! Please wait a moment...
(You could hang up the call from PSTN if you get this message)

Frequency 1 : 473
Frequency 2 (2623) is more than 1000, please ignore it.
0. 25sec on 0.25sec off

tone 4 473 473 8 8 25 25 1023 1023
(Please configure the high and low frequency as the same value if you just get a singal frequency)

tone –print

Disconnect tone 1 paramter
- Frequency high : 620
- frequency low : 480
- frequency high level : 8
- frequency low level : 8
- Tone1 on : 25
- Tone1 off : 25
- Tone2 on : 1023
- Tone2 off : 1023

Disconnect tone 2 paramter
- Frequency high : 450
- frequency low : 0
- frequency high level : 8
- frequency low level : 0
- Tone1 on : 35
- Tone1 off : 35
- Tone2 on : 1023
- Tone2 off : 1023

Disconnect tone 3 paramter
- Frequency high : 620
- frequency low : 480
- frequency high level : 8
- frequency low level : 8
- Tone1 on : 50
- Tone1 off : 50
- Tone2 on : 1023
- Tone2 off : 1023

Disconnect tone 4 paramter
- Frequency high : 473
- frequency low : 473
- frequency high level : 8
- frequency low level : 8
- Tone1 on : 25
- Tone1 off : 25
- Tone2 on : 50
- Tone2 off : 50

(Confirm the values is correct or not)

(Key in the commit and reboot command if you finish the procedures as above)

4.2.22[support]

Special Voice function support manipulation

Special Voice function support manipulation
Usage:
support [-t38 enable][-t38ecm enable][-faxrdd digits]
support -print
-t38 T.38(FAX) enabled/disabled.
-t38ecm T.38(FAX) ECM enabled/disabled.
-faxrdd FAX redundancy depth(0 ~ 2).
Example:
support -t38 1
support -t38ecm 1
support -faxrdd 1

usr/config$

Parameter Usages:

-t38 Enable or disable T.38 fax ability. The function will automatically switch codec (G.723 or G.729a) to T.38 when FAX signal is detected.

-t38ecm Enable or disable t38ecm function. The function is support the error correction in the high-speed fax mode.

-faxrdd Set Fax redundancy depth. User can increase FAX redundancy depth when network traffic is heavy. For example, if user set fax redundancy as 2, Gateway will resend fax packets every 2 packets.

4.2.23[tos]

IP Packet ToS(type of Service)/Differentiated Service configuration.

---

usr/configtos

IP Packet ToS(type of Service)/Differentiated Service configuration
Usage:
tos [-rtptype dscp]
tos [-sigtype dscp]
tos -print
[-rtpreliab mode]
tos -print

Example:
tos -rtptype 7 -sigtype 0

Parameter Usages:
-rtptype the packages of voice
-sigtype the package of call signal

Note:
The value of rtptype and sigtype is from 0 to 63.
It's working if it supported by your network.

4.2.24 [pt]

RTP payload type configuration and information

---

usr/config$ pt

RTP payload type configuration and information
Usage:
pt -print Display the RTP payload type information
-rfc2833 Configurate the DTMF RFC2833 payload type
-dtmf Configurate the DTMF payload type
-fax Configurate the FAX payload type
-faxbypass Configurate the FAX ByPass payload type
-modembypass Configurate the MODEM ByPass payload type
-redundancy Configurate the Redundancy payload type
-modemrelay Configurate the MODEM Relay payload type

Example:
pt -rfc2833 96 -fax 101

```
usr/config$
```

4.2.25 [rom]

ROM file information and firmware upgrade function.

```
---------------------------------------------------------------------------------------------
usr/config$ rom

ROM files updating commands
Usage:
rom [-print][-app][-boot][-dsptest][-dspcore][-dspapp][-greet][-askpin]
    -s TFTP/FTP server ip -f filename
rom -print
    -print       show versions of rom files. (optional)
    -app          update main application code(optional)
    -boot         update main boot code(optional)
    -dsptest     update DSP testing code(optional)
    -dspcore     update DSP kernel code(optional)
    -dspapp       update DSP application code(optional)
    -greeting    update greeting voice file(optional)
    -askpin       update ask pin code voice file(optional)
    -s            IP address of TFTP/FTP server (mandatory)
    -f            file name(mandatory)
    -method          download  via  TFTP/FTP  (TFTP:  mode=0,  FTP:
mode=1)
    -ftp          specify username and password for FTP
    -server       specify EMS Server IP address
    -id           specify EMS Server ID
    -pwd          specify EMS Server password
    -emstime      specify EMS cycle time
Note:
    This command can run select one option in 'app', 'boot',
    , 'dsptest', 'dspcore', and 'dspapp'.
Example:
    rom -method 1
    rom -ftp vwusr vwusr
    rom -app -s 192.168.4.101 -f app.bin
```

usr/config$

Parameter Usages:

-print show versions of all rom files

-app, boot, dsptest, dspcore, dspapp, greeting, askpin

To update main Application program code, Boot code, DSP testing code, DSP kernel code, DSP application code, greeting file, askpin file.

Note: To set mac address please key in command setmac:(when key in MAC address ,press enter each time after key in two characters)

Please get the MAC address from the bottom of your FXO VOIP units.

---

usr/config$ setmac

- enter mac address (xxxxxxxxxxxx):

0001a8002baa

- the mac address is 00 01 a8 00 2b aa

- if mac address is correct,please press 'y' to

setup configuration,else press 'n' to continue

y

-greeting The greeting file can be updated by users. The attributes of sound file should complied to: μ-law, 8000 Hz , 8 bit, Mono, 7 kb/s

-askpin update ASKPIN sound file. This is the greeting sound that when asking for pincode.

-s to specify TFTP server's IP address when upgrading ROM files.

-f to specify the target file name, which will replace the old one.

-method to decide using TFTP or FTP as file transfer server. "0" stands for TFTP, while "1" stands for FTP.

-ftp if users choose FTP in above item, it is necessary to specify pre-defined username and password when upgrading files.

4.2.26 [passwd]

For security concern, users have to input the password before entering configuration mode.

---

usr/config$ passwd

Password setting information and configuration
Usage:
passwd -set Loginname Password
passwd -clean
Note:
1. Loginname can be only 'root' or 'administrator'
2. passwd -clean will clear all passwd stored in flash,
please use it with care.
Example:
passwd -set root Your_Passwd_Setting

usr/config$

Parameter Usages:
-set
(***passwd –set "login name" "password"***)
Note : "login name" can be "***root***" or "***administrator***" only. "root" and "administrator" have the same authorization, except some commands that can be executed by "root" only – "***passwd –clean***", "***rom –boot***", "***rom –bot2m***" and "***flash –clean***".

4.2.27 [auth]

For security concern, the "root" user can customize some configurable items for "administrator" user.

---

usr/config$ auth

Root control what command administrator can use.
Usage:
auth -print Display auth switch configuration.
Use item name to do config name (0=Disable, 1=Enabled).
Example: auth -ifaddr 1

usr/config$

Parameter Usages:

-print Display the configurable items for “administrator” user.

-“item name”Assign the configurable item for “administrator” user.

Note : For example, you can set below authorization for root user.

```
usr/config$ auth -ifaddr 1
usr/config$ auth -sip 1
usr/config$ auth -voice 1
```

Now the Administrator can use these command which Root user assign to them.